Session 3aAB

Animal Bioacoustics: Passive and Active Marine Bioacoustics

Juliette W. Ioup, Chair
Univ. of New Orleans, Dept. of Physics, New Orleans, LA 70148

Chair's Introduction—8:00

Contributed Papers

8:05

3aAB1. A simple ocean noise exposure metric based on naturally occurring noise levels and biological thresholds. Michael Stocker, Tom Reuterdahl, Libbie Horn, and Gail Hurley (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938)

Anthropogenic noise is compromising the habitat for marine mammals, fish, and, potentially, other marine organisms. Determining acceptable thresholds is confounded by the fact that marine animals have adapted to some exceedingly loud naturally occurring sounds, whereas exposure to certain anthropogenic noises at equivalent or lower amplitudes causes harm. It is clear that mitigation levels cannot be established by signal amplitude alone. This proposed metric helps establish exposure levels based on broadband and temporal representation of a subject noise compared to a set of spectral curves based on ambient noise levels and biological thresholds.

8:20

3aAB2. Ocean acoustic effects of explosions on land: Evaluation of Cook Inlet beluga whale habitability. Sara K. Tremblay (Animal Sci. Dept., Univ. of Connecticut, Storrs, CT 06269), Thomas S. Anderson (U.S. Army Corps of Engineers, ERDC, Hanover, NH 03755), Erin C. Pettit (U.S. Army Corps of Engineers, CRREL, Fairbanks, AK 99703), Peter M. Scheifele (Univ. of Cincinnati, OH 45221), Gopu R. Potty, and James H. Miller (Univ. of Rhode Island, Narragansett, RI 02882)

The Eagle River Flats is an impact region for artillery at Fort Richardson, Alaska. Adjacent to the Flats is the Knik Arm of Cook Inlet which is the habitat for a distinct population of beluga whales (Delphinapterus leucas). In order to assess the effects of 155 mm artillery explosions on the habitat of these whales, a series of 6.8 kg C4 plastique charges were detonated on land 500 meters from the waters edge. In addition to land seismic and acoustic arrays, hydrophones were deployed in the Knik Arm at high and low tide. This paper discusses the ocean acoustic measurements. The received signal 30 meters from the shore in water depths of 8 meters was more intense at high tide, with broadband peak levels of approximately 180 dB *re* 1 microPa. The dominant frequency was about 20 Hz and most of the received acoustic energy was below 500 Hz. The geology and oceanography of the area were used to model the acoustic time series. Modeled and measured time series are compared to validate the geophysical model and provide estimates of peak pressure and energy flux density over the near shore habitat. [Work supported by U.S. Army Corps of Engineers, CRREL.]

8:35

3aAB3. Acoustic environments of bottlenose dolphins (*Tursiops truncatus*) in the Big Bend region of Florida. Athena Rycyk and Douglas Nowacek (Florida State Univ. Dept. of Oceanogr. 105 N. Woodward Ave., P.O. 3064320 Tallahassee, FL 32306-4320, Rycyk@ocean.fsu.edu)

The researchers examined the acoustic environments of coastal bottlenose dolphins (Tursiops truncatus) in the Big Bend region of Florida, one of the most pristine coastal environments in the state, by means of remote autonomous acoustic recorders deployed at eight sites. The rates and types of dolphin vocalizations (whistles, echolocation, burst-pulse calls, and pops) differed between the eight sites. Dolphins were not the only source of noise in their marine habitat; other biological and anthropogenic sounds occurred. Each site was found to have a unique soundscape with respect to dolphin, fish, snapping shrimp, and anthropogenic sound. Toadfish (Opsanus beta), silver perch (Bairdiella chrysoura), and sea catfish (Arius felis) were the only identifiable fish species to produce sound and each caused notable increases in sound levels at low frequencies where they vocalized. Locations exhibited different time peaks in fish produced sounds. Overall, anthropogenic noise was uncommon and only found in the form of boat noise. Surprisingly, dolphin vocalizations were not found at all locations only at Alligator Harbor, Dog Island, Carabelle River, and West Pass. The lack of dolphin vocalizations does not imply dolphins were not present in these regions only that no dolphins were vocalizing at the time of recordings.

8:50

3aAB4. Preliminary acoustic data analysis from Littoral Acoustic Demonstration Center 2007 (LADC07) passive acoustic experiment. Natalia A. Sidorovskaia (Dept. of Phys., UL Lafayette, UL BOX 44210, Lafayette, LA 70504-4210), George E. Ioup, Juliette W. Ioup, Arslan M. Tashmukhambetov (Univ. of New Orleans, New Orleans, LA 70148), Grayson H. Rayborn (Univ. of Southern Mississippi, Hattiesburg, MS 39406), Joal J. Newcomb, Robert Field, and Guy Norton (Naval Res. Lab., Stennis Space Ctr., MS 39529-5004)

The Littoral Acoustic Demonstration Center conducted a passive acoustic experiment in the Northern Gulf of Mexico, about 120 nmi south of the Mississippi delta between July 5 and July 17, 2007. The primary objective of the experiment was to make probably the first passive acoustic recordings of beaked whales in the Gulf of Mexico. Six environmental acoustic recording system (EARS) buoys were deployed in two triangular configurations in about 1,500 m water with the hydrophones located at a 1,000 m depth. The distance between the two triangles was about 20 nmi. Each EARS buoy recorded in the frequency band of 100 Hz–96 kHz for ten consecutive days, with a total data amount collected of about 3 Tbytes. More than ten different marine mammal species were identified in the vicinity of the buoys during visual observations, which supported the experiment including three different types of beaked whales. Preliminary environmental and acoustic data analysis, including average noise curves and different species identification by using newly developed frequency-domain detectors, is presented. Assessment of the density of recorded acoustic phonations of different species is discussed. [Research supported through SPAWAR PMW-180 PE63207N.]

9:05

3aAB5. Sperm whale tracking in the Bahamas: One hydrophone, three dimensions. Christopher O. Tiemann (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, tiemann@arlut.utexas.edu)

A passive acoustic localization method for tracking the movement of a clicking sperm whale in three dimensions using data from just one hydrophone is demonstrated, using data made available for the 3rd International

Workshop on Detection and Classification of Marine Mammals. One recording contains sperm whale clicks recorded on a bottom-mounted hydrophone on a steep slope of the Navy's AUTEC test range. When the direct-path acoustic ray arrivals from several clicks are time-aligned, persistent associated multipath arrivals of reflected ray paths can be identified for each click event and used for localization. Although the use of multipath arrival information is a standard procedure for range-depth tracking, a three-dimensional estimate of whale position can be obtained from the same multipath information with knowledge of an azimuthally-dependent environment, relative to the receiver. In this case, azimuthal distinction arises from varied bathymetry. Multipath arrival patterns are matched to unique range-, depth-, and azimuth-dependent modeled arrival patterns to make an estimate of whale location. A three-dimensional whale track in range, depth, and bearing from the fixed hydrophone will be presented.

9:20

3aAB6. Tracking baleen whales using the relative relocation method. Catherine L. Berchok, Gerald L. D'Spain, and John A. Hildebrand (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA 93940-0701)

Passive acoustic tracking of baleen whales is typically accomplished using time-of-arrival difference measurements from widely-separated acoustic sensor arrays. Because these arrays are large, the propagation conditions along the paths from the source to each of the receivers can be quite different, causing the received signal characteristics to vary among arrivals from the same vocalization. These signal variations introduce uncertainty in the time-of-arrival difference measurements, which in turn creates scatter in the localizations, obscuring the tracks of the individual whales. The relative relocation method, developed in the seismic community, provides a means of reducing scatter in the localizations by using the differences in time-of-arrival differences to adjust the locations of signals relative to one another. The method, as used for earthquake localizations, will be described, then applied to a set of fin whale vocalizations recorded on an ocean bottom seismometer array deployed off southern California. The results provide better defined tracks with less scatter than traditional localization techniques.

9:35

3aAB7. Whale tracking underwater: High frequency acoustic pingers and the instrumented tag (DTAG). Val E. Schmidt, Thomas C. Weber, Colin Ware, Roland Arsenault (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824), David Wiley (Natl. Oceanograph. and Atmospheric Assoc., Situate, MA), Mark P. Johnson, Erik Dawe (Woods Hole Oceanograph. Inst., Woods Hole, MA), and Ari S. Friedlaender (Duke Univ., Beaufort, NC)

Since 2004, scientists have been tagging and tracking humpback whales in Stellwagen Bank National Marine Sanctuary to better understand their behavior. Stellwagen Bank is a shoal area east of Boston and north of Cape Cod, MA where many species of baleen whale feed during the summer months. Instrumented tags (DTAGs) are suction-cupped to the whales back from a RHIB. DTAGS, developed at WHOI, record whale pitch, roll, and heading, 3-D acceleration, depth, and sound for up to 20 h. A pseudotrack for the tagged whale can be generated using visual fixes at the surface and dead-reckoning while the whale is underwater. During extended dives, the solution is expected to exhibit substantial drift, placing limits on the ability to understand feeding behavior, mother-calf interactions, etc. In order to develop higher accuracy whale tracks, three GPS-positioned high-frequency (25–32 kHz) acoustic pingers were deployed around tagged animals in July 2007. The pingers produce time-encoded pulses from known positions, which are recorded along with whale vocalizations and ambient noise on the whale tag. Pulse arrival times from each pinger are converted into ranges from the known pinger locations to generate an underwater whale track. Results from this work will be presented.

9:50

3aAB8. Acoustic identification of beaked and sperm whales. George E. Ioup, Juliette W. Ioup, Lisa A. Pflug, Arslan M. Tashmukhambetov (Dept. of Phys., Univ. of New Orleans, New Orleans, LA 70148), and Natalia A. Sidorovskaia (Univ. of Lousiana at Lafayette, Lafayette, LA 70504)

Identification of individual marine mammals acoustically was initially motivated in this research by spectrograms of the littoral acoustic demonstration center Northern Gulf of Mexico data containing sperm whale click codas, which showed that clicks in a coda have a spectral pattern that persists across all the clicks in that coda. The hypothesis is that each coda originates from a single whale, and all codas with similar properties come from the same whale. Self-organizing maps (SOMs) are used to compare and classify the time series, Fourier transforms, and wavelet transforms of each coda in order to determine how many whales are present. The results show that SOMs have promise for classifying underwater acoustic coda signals from sperm whales. A similarity measure has been applied to both coda and echolocation clicks and has shown some success in associating both types of clicks with individuals. Other forms of cluster analysis are also considered. Progress in the classification of sperm whale clicks has motivated the application of similar analysis to beaked whale echolocation clicks. Time-frequency plots show interesting details. Preliminary results for beaked whale click analysis are presented. [Research supported by ONR.]

10:05–10:35 Break

10:35

3aAB9. Spawning behavior and spatial distribution of Atlantic herring on Georges Bank revealed by ocean acoustics waveguide remote sensing. Zheng Gong, Daniel Cocuzzo, Mark Andrews, Purnima Ratilal (Northeastern Univ., 360 Huntington Ave., Boston, MA 02115), Srinivasan Jagannathan, Nicholas Makris (Massachussetts Inst. of Technol., Cambridge, MA 02139), Hector Pena, Ruben Patel (Inst. of Marine Res., Bergen, Norway), and J. Michael Jech (NOAA/Northeast Fisheries Sci. Ctr., Woods Hole, MA 02543)

An ocean acoustics waveguide remote sensing (OAWRS) system was deployed in the Gulf of Maine, near Georges Bank to image Atlantic herring and other fish population from September–October 2006. OAWRS provides spatially unaliased imaging of herring over wide areas, spannning over 100 km diam. Migration and spawning behavior of Atlantic herring were observed using OAWRS over several diurnal periods, including massive movements on and off the bank to spawn. Measurements made simultaneously with a conventional fish-finding echosounder (CFFS) and a multibeam sonar provide the depth distribution and local 3-D morphology, respectively, of the herring schools in the water column. Concurrent trawl surveys provide identification of the fish species. Measurements made by OAWRS and CFFS systems are highly correlated. Examples will be provided of the co-registration between the two systems over a one-week period. Calibration of the OAWRS system using CFFS estimates of fish population densities, along with a full-field scattering model that takes into account both coherent and incoherent scattering from a fish group, is discussed. Resonance scattering behavior of herring is observed in the OAWRS system with significant changes in scattering amplitude over the 300 Hz to 1.5 kHz frequency range of the OAWRS system.

10:50

3aAB10. Multibeam and single-beam sonar observations of Atlantic herring in the Gulf of Maine. Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824), Hector Pena (Inst. of Marine Res., Bergen, Norway), and J. Michael Jech (Northeast Fisheries Sci. Ctr., Woods Hole, MA)

In September 2006, several schools of Atlantic herring were simultaneously imaged with a Reson 7125 SeaBat multibeam sonar (400 kHz) and a Simrad EK60 Scientific echo sounder (38 kHz, 120 kHz, and 200 kHz) on Georges Bank in the Gulf of Maine. One school was imaged five

separate times over the span of one hour. These two sonar systems, along with pelagic trawl catch data collected on a separate ship, provided a synoptic view of the fish school as it changed depth and size, fragmented, and eventually dispersed. Of particular interest is the combination of data from the two sonars. The multifrequency, split-beam EK60 provided estimates of the fish school density directly under the ship, but was not capable of accurately extrapolating these measurements to the entire fish school, which was largely out of the EK60 field of view. The multibeam sonar was generally able to image the entire fish school and provided useful information on school volume, fragmentation, and general school morphology, but was more difficult to use for quantifiable measurements, due to the unknown angle of ensonfication of the fish in the outer beams.

11:05

3aAB11. Measurement of material properties of two gelatinous, coastal zooplankton. Joy N. Smith (Dept. of Marine Sci., Coastal Carolina Univ., Conway, SC 29528) and Joseph D. Warren (Stony Brook Univ., Southampton, NY 11968, joe.warren@stonybrook.edu)

Acoustic scattering models for zooplankton are often used to transform acoustic data (backscattered energy) into biological units (number or size of animals). In order for these models to be used successfully, accurate descriptions of the animals themselves are required including: Species, size, orientation, and their material properties, specifically the density and sound speed of the organism relative to their surroundings. Measurements were made of the density and sound speed of two coastal, gelatinous zooplankton (ctenophore *(Mnemiopsis leidyi)*, lion's mane jellyfish *(Cyanea capillata)*) found in the waters around Long Island, New York. Two different methods were used to determine the density of these zooplankton, while an acoustic-travel-time approach was used to measure the sound speed of these animals. Mean animal density was related to animal size for both species and varied between individuals. Values of density and sound speed are reported and their implications with regard to the use of acoustic surveys to assess gelatinous zooplankton are discussed. [Work supported by NSF.]

11:20

3aAB12. Effects of otolith geometry on steady streaming flows in the fish ear. Charlotte W. Kotas, Peter H. Rogers, and Minami Yoda (Georgia Inst. of Technol., Atlanta, GA 30332-0405, charlotte.kotas@gatech.edu)

The dense, bony otolith contained in the fish ear oscillates relative to its fluidic surroundings in the presence of a sound wave. How the otolith actually transduces this acoustically induced fluid motion into the hair cell displacements that the fish hears is not fully understood, however. Nevertheless, it is likely that the complicated geometry of the otolith, including the groove (sulcus) where most of the hair cells are found, contributes in some fashion to shaping the flow patterns that excite the hair cells. The effect of these grooves on the induced steady streaming flows was studied experimentally using both oscillating grooved spheroids and scaled models of actual otolith sulci to simulate acoustically induced motions of the otolith for oscillation orientation angles ranging from 0 deg to 90 deg. Particle-image velocimetry and flow visualization results obtained from images phase-locked to the oscillations are presented for normalized oscillation amplitudes, $\epsilon \equiv s/L = 0.05-0.2$ and Reynolds number, $\mathrm{Re} \equiv \omega L^2/\nu \approx 1-10^2$, where s is the oscillation amplitude, ω is the oscillation frequency, L is a typical otolith length scale, and ν is the fluid kinematic viscosity. The effect of phase on these data was also studied. [Work supported by ONR.]

THURSDAY MORNING, 29 NOVEMBER 2007 GRAND COUTEAU, 8:00 TO 10:20 A.M.

Session 3aAO

Acoustical Oceanography: Acoustic Effects of Internal Waves and Other Finescale Oceanography

James F. Lynch, Chair
Woods Hole Oceanographic Inst., Bigelow Bldg., Woods Hole, MA 02543

Chair's Introduction—8:00

Contributed Papers

8:05

3aAO1. A feed-back algorithm using ocean acoustic modeling to improve shallow water soliton simulations. Stanley A. Chin-Bing, Alex C. Warn-Varnas, and David B. King (Naval Res. Lab., Stennis Space Ctr., MS 39529, chinbing@nrlssc.navy.mil)

Ocean solitons can affect the acoustic signals that travel through them. The effects can range from slight to severe. Computer simulations can be used to predict the large-scale effects on the acoustic signal. The typical sequence of events requires that an ocean model be initialized by tidal velocity and used to estimate the changes in the environmental parameters due to soliton creation and propagation. Changes in the environment are used to calculate the related changes in the ocean sound speed field. The last step is to run an ocean acoustic computer model to predict the changes in the acoustic signal. Often, the tidal velocity is not precisely known and assumptions have to be made. Any variations in the tidal velocity require the time consuming sequence to be repeated. Recently, we have demonstrated a way of estimating the soliton structure that could affect the acoustic signal. This estimation is made before any ocean model simulation is made. This can greatly reduce the number of computer simulations, since only ocean model simulations are made for those conditions that might significantly affect the acoustic signal. Examples will be shown that illustrate this acoustic "feedback" method. [Work supported by the NRL Base program.]

8:20

3aAO2. Acoustic cost functions for autonomous adaptive sampling. Kevin D. Heaney (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039)

Recent advances in ocean modeling have provided the opportunity to forecast ocean conditions out to 48 h. Accuracy of these models is determined primarily by the input conditions (boundary, as well as internal state). An automated algorithm for adaptively sampling the ocean environment has been developed with the goal of providing sampling paths for regions where input drives the model solutions the most. In this paper, a set of acoustic cost functions is presented, which should indicate regions where acoustic propagation is most sensitive to model uncertainty and/or model-predicted temporal variability. The four cost functions explored will be: Matched mode coherence, mode coupling coefficients, transmission loss variability, and detection range coverage. This technique will be ap-

plied to the data-assimilated model results from the Shallow Water 2006 experiment, where oceanographic measurements consisted of a large array of thermistors, six gliders operating continuously, and a ship towed scanfish. Acoustic measurements included fixed source/receivers, AUV mounted sources, as well as sonobuoy receivers. Sampling guidance based upon the acoustic cost functions will be used in an observational systems simulation experiment (OSSE), in collaboration with Pierre Lermusiaux and his group at MIT. For the ocean modeling, data assimilation and sampling suggestions see http://modelseas.mit.edu.

8:35

3aAO3. Tidal time-scale variations of sound propagation in the Hudson River Estuary. Sreeram Radhakrishnan, Alexander Sutin, and Alan Blumberg (Ctr. for Maritime Systems, Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030)

The nowcast and forecast of sound propagation in an urban estuary is important for the development of acoustic surveillance systems. Sound propagation in an estuary is highly affected by the temporal and spatial variability of salinity and temperature due to tides, freshwater inflows, winds, etc. Estuarine processes are analyzed here focusing on the formation and breakdown of the salinity and temperature stratification and their influence on sound attenuation. Transmission loss (TL) variability was determined first using real-time data collected in the Hudson River near Hoboken, New Jersey. For TL forecasting, 24-h forecasts of salinity and temperature distributions from a high-resolution New York harbor observing and prediction system (NYHOPS) developed at Stevens were used (www.stevens.edu/maritimeforecast/). These forecasts provided the basis for calculating sound speed profiles. A parabolic equation based acoustic model was implemented to simulate the acoustic field structure. Real-time TL measurements were made by transmitting sweep frequency signals (1–100 kHz) to a distance of 175 m. The observed tidal time-scale TL variation of 10 dB was in good agreement with the model calculations. The results indicate that TL can be predicted within the context of the NYHOPS system. [Work supported by ONR Project No. N00014-05-1-0632: Navy Force Protection Technology Assessment Project.]

8:50

3aAO4. Applying data nullspace projection method on matched-field source localization in a random internal wave field. Ying-Tsong Lin, James F. Lynch, and Arthur E. Newhall (Woods Hole Oceanogr. Inst., Woods Hole, MA 02543)

In a random internal wave field where temporal varying water-column soundspeed profiles are very difficult to measure, matched-field source localization methods can be greatly degraded due to water-column model mismatch. In this paper, the data nullspace projection is applied to this source localization technique to reduce the effect of unknown water-column soundspeed variations. The basis of this method is to project the acoustic signal onto a data nullspace that is insensitive to water-column soundspeed fluctuations. This method only requires the mean and the second-order statistics of temporal varying water-column soundspeed profiles to calculate soundspeed empirical-orthogonal-functions. It does not require measurements of the exact soundspeed field, i.e., each snapshot, for the matched-field processing. In a simulation test case, a linear wave model is used to generate a random internal wave field, an acoustic source continuously transmits a single frequency tone, and the matched-field processing is implemented with the signal received on both a VLA and a HLA, respectively, for localizing the source position. The simulation results show that applying the data nullspace projection method can dramatically improve the robustness and accuracy of the matched-field source localization, resulting in a random internal wave field.

9:05

3aAO5. Coherence function of a sound field in an oceanic waveguide with horizontally isotropic random inhomogeneities. Alexander G. Voronovich and Vladimir E. Ostashev (NOAA/Earth System Res. Lab., 325 Broadway, Boulder, CO 80303)

Closed equations for the coherence function of a monochromatic sound field propagating in a 3-D oceanic waveguide with random inhomogeneities were derived in many works, e.g., see [A. Voronovich and V. Ostashev, J. Acoust. Soc. Am. **119**, 1406–1419 (2006)] and references therein. However, due to high dimensions of matrices appearing in these equations, their numerical solution still remains problematic. In this paper, a closed equation for the coherence function due to a point omnidirectional source in an oceanic waveguide is derived for the case of random inhomogeneities, which are statically isotropic in a horizontal plane. Due to cylindrical symmetry of the problem, dimensions of the matrices appearing in the derived equation are significantly reduced, as compared to a general case. This makes the equation for the coherence function readily amenable for numerical calculations. Using this equation, the effects of internal waves with the Garret-Munk spectrum on the coherence function of a sound field propagating in an oceanic waveguide are studied numerically.

9:20

3aAO6. Effect of ocean internal waves on the interference component of the acoustic field in the Long-range Ocean Acoustic Propagation Experiment. Natalie S. Grigorieva, Gregory M. Fridman (Dept. of Appl. Math., St. Petersburg State Marine Tech. Univ., 3 Lotsmanskaya Str., St. Petersburg, 190008, Russia, nsgrig@natalie.spb.su), James Mercer, Rex Andrew, Bruce Howe, Michael Wolfson (Univ. of Washington, Seattle, WA 98105), and John Colosi (Naval Postgrad. School, Monterey, CA 93943)

The propagation of energy along the sound-channel axis cannot be described in terms of geometrical acoustics because of the presence of cusped caustics repeatedly along the axis. In neighborhoods of these cusped caustics a very complicated interference pattern is observed. Neighborhoods of interference grow with range and at long ranges they overlap. This results in the formation of a complex interference wave-the axial wave-that propagates along the sound-channel axis like a wave belonging to a crescendo of near-axial arrivals. In this paper, the axial wave is simulated for the LOAPEX CTD data measured at seven different ranges from the vertical line array. A signal with the center frequency of 75 Hz and 37.5-Hz bandwidth is used for computations. This signal well approximates one transmitted m-sequence in the LOAPEX experiment. The effect of environmental variability, induced by internal waves, on the axial wave is studied. The sound-speed fluctuations caused by ocean internal waves are obtained with the use of the buoyancy frequency profile measured in the LOAPEX. Calculations are based on the integral representation of the axial wave in a local coordinate system introduced in the vicinity of the range-variable sound-channel axis. [Work supported by ONR.]

9:35

3aAO7. Horizontal coherence of low-frequency shallow water signals. Jon M. Collis (Boston Univ., Boston, MA, jcollis@whoi.edu), Timothy F. Duda, James F. Lynch, Arthur E. Newhall, Keith von der Heydt (Woods Hole Oceanograph. Inst., Woods Hole, MA 02543), Harry DeFerrari, and Hien Ba Nguyen (Univ. of Miami, Miami, FL 33149)

Signals from moored sources at a range of approximately 20 km distant from a fixed L-shaped receiver array (joint horizontal and vertical line array) on the New Jersey area shelf have been analyzed to determine horizontal coherence properties of the signals. The signals, from 100 Hz to 800 Hz, do not arrive broadside to the horizontal array. Subsequently they show phase and amplitude fluctuations attributable to interference within a field having an azimuthally uniform normal mode structure, and azimuthal variations of the modal structure. Synthesis of an interference pattern using the acoustic field measured with the vertical array is used to distinguish between the two effects. High temporal variability of the inferred

3a THU. AM

azimuthal variation of modal content correlates with internal wave activity observed along the path. Horizontal coherence lengths are often of the order of ten acoustic wavelengths. [Work supported by ONR.]

9:50

3aAO8. Effects of offshore mesoscale eddies and fronts on inshore shallow water acoustic propagation. Harry Deferrari (Div. of Appl. Marine Phys., Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149)

Shallow water shelf areas inside of western boundary currents have two distinctly different ocean acoustic environments determined by the type of front that separates deep offshore from shallow inshore. Prograde fronts allow for the ducting of offshore internal waves up onto the shelf as the main source the sound speed fluctuations over the internal wave band. Retrograde fronts block the propagation of offshore internal waves setting up stability conditions that allow for propagation of locally generated, large amplitude non-linear solitary waves as the major source of sound speed variability. Here, acoustic propagation is examined for both environments with data from two similar fixed system propagation experiments, one for the prograde environment off the coast of south Florida near the site off the Acoustic Observatory, and the second for the retrograde front environment of the Mid-Atlantic Bight the SWO6 site. Offshore mesoscale features of fronts and eddies are shown to determine mean sound speed profiles and the energy of the internal wave fields. In turn, intensity fluctuations and temporal coherencies of broadband acoustic signals over several octaves are observed to vary with variations of the sound speed. For some locations, observations of mesoscale features alone can predict sonar performance.

10:05

3aAO9. Observations of temporal coherence for broad band acoustic transmission in shallow water. Harry Deferrari (Univ. of Miami, 4600 Rickenbacker Cswy. Miami FL, 33149)

The temporal coherencies of broadband signals are computed with data from fixed system measurements at two experimental sites. The acoustic source for both measurements was the Miami Sound Machine (MSM) which transmits continuous signals at 6 carrier frequencies from 100 to 3200 Hz in octave steps. Each signal has a bandwidth of 25% of carrier. The bandwidth of the acoustic signal allows the identification of paths by arrival time, so that coherence can be estimated for individual paths and for multipath groupings. The relation between internal wave energy and acoustic signal coherence times is explored. Internal wave energy is estimated with moored T-D instruments along the path of propagation. The sound speed variability resulting from internal waves at the two sites is markedly different: one is driven by locally generated nonlinear solitary waves and, the other, by offshore internal waves that propagation into the shallow water. The low frequency signals are directly modulated in amplitude and phase at the period of the wave passing the site, whereas, the higher frequencies have saturated statistics without evidence of direct modulation. A puzzling observation is that coherence times do not change much while the IW energy increases by 2 orders of magnitude.

THURSDAY MORNING, 29 NOVEMBER 2007 — MAUREPAS, 8:15 TO 11:45 A.M.

Session 3aBB

Biomedical Ultrasound/Bioresponse to Vibration: Current Topics in Diagnostic and Therapeutic Ultrasound

Jeffrey A. Ketterling, Chair

Riverside Research Inst., 156 William St., New York, NY 10038-2609

Contributed Papers

8:15

3aBB1. Optimized translation of microbubbles driven by acoustic fields. Jean O. Toilliez and Andrew J. Szeri (Dept. of Mech. Eng., Univ. of California, 6112 Etcheverry Hall, Berkeley, CA 94720-1740, toilliez@me.berkeley.edu)

The problem of a single acoustically driven bubble translating unsteadily in a fluid is considered. The investigation of the translation equation allows for identifying the inverse Reynolds number as small perturbation parameter. The objective is to obtain a closed-form, leading order solution for the translation of the bubble, assuming nonlinear radial oscillations and a pressure field as the forcing term. The result is the ability to predict and explicitly understand the rapid and slow transients of bubble displacement, which is proportional to the average acoustic radiation force. The periodic attractor of the Raleigh-Plesset equation serves as basis for an optimal acoustic forcing designed to achieve maximized bubble translation over one dimensionless period. At moderate acoustic intensity, maximizing the radial variance, thereby enhancing bubble collapse, leads to displacement many times larger than the case of purely sinusoidal forcing. The survey covers a wide spectrum of driving ratios and bubble diameters physically relevant to biomedical applications. Shape stability issues are considered. Together, these results suggest new ways to predict some of the direct and indirect effects of the acoustic radiation force in biomedical applications: e.g., targeted drug delivery, selective bubble driving and accumulation. [Work supported by NASA Microgravity Fluid Physics Program.]

8:30

3aBB2. Growth and dissolution of a contrast microbubble: Effects of encapsulation. Kausik Sarkar and Pankaj Jain (Mech. Eng., Univ. of Delaware, Newark, DE 19716)

Micron-size gas bubbles are intravenously injected into patients body at the time of ultrasound imaging to improve image contrast. The bubbles are encapsulated by a thin layer (4–10nm) of protein, lipids and other surface active materials, to prevent their premature dissolution in the blood. A model will be presented that describes the dissolution of a microbubble accounting for the effects of encapsulation. The encapsulation hinders the permeability of the gas-liquid surface and its elasticity balances the surface tension induced stress. Both these effects will be explicitly modeled. The model behavior will be discussed for variations of the material parameters and conditions (encapsulation permeability and elasticity, mole fraction of the osmotic agent and liquid saturation). The encapsulation significantly affects the bubble growth and dissolution including their time scales.

8:45

3aBB3. Echogenic liposomes loaded with recombinant tissue-type plasminogen activator (rt-PA) for image-guided, ultrasound-triggered drug release. Denise A. B. Smith, Sampada Vaidya, Jonathan A. Kopechek, Kathryn E. Hitchcock (Dept. of Biomed. Eng, Univ. of Cincinnati, Medical Sci. Bldg, Rm. 6152, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, smitdn@email.uc.edu), Shaoling L. Huang, David D. McPherson (Univ. of Texas Health Sci. Ctr., Houston, TX 77030), and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH 45267-0586)

A recently developed ultrasound contrast agent, rt-PA-loaded echogenic liposomes (TELIP), was assessed in vitro using a clinical diagnostic ultrasound scanner (Philips HDI 5000) equipped with a linear array (L12-5). The stability and echogenicity of static TELIP suspensions were determined using 4.5-MHz harmonic B-mode pulses ($P_r=120$ kPa; MI $=0.04$) in an anechoic chamber. An in vitro flow phantom with a flow rate of 5 ml/min at 37 °C was also used to assess TELIP for ultrasonically-triggered drug release. TELIP samples were exposed to: (1) Fundamental 6.9-MHz B-mode pulses ($P_r=600$ kPa; MI$=0.04$) where diffusion of gas out of the liposomes occurs over 60 min, or (2) 6.0-MHz color Doppler pulses (PD$=3.33$ μs, PRF$=1$ kHz) at two exposure levels, 0.8 MPa (MI$=0.22$) for which acoustically driven diffusion was evident or 2.6 MPa (MI$=0.7$), for which rapid fragmentation was confirmed. Exposure of TELIP to Triton-X, a nonionic detergent, served as a positive control for drug release. Release of rt-PA for each ultrasound exposure protocol was assayed spectrophotometrically (Shimadzu UV-1700). The thrombolytic drug remained associated with the lipid bilayer when exposed to B-mode pulses over time and was released when exposed to color Doppler pulses. [This work was supported by NIH 1RO1 NS047603 and NIH 1R01 HL074002.]

9:00

3aBB4. Ultrasound-mediated release of calcein from echogenic liposomes. Jonathan A. Kopechek, Stephen M. Chrzanowski, Denise A. B. Smith, Whitney B. Gaskins (Dept. of Biomed. Eng., Univ. of Cincinnati, Medical Sci. Bldg., Rm. 6152, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, kopechja@uc.edu), Todd A. Abruzzo (Univ. of Cincinnati, Cincinnati, OH 45267), Shaoling L. Huang, David D. McPherson (Univ. of Texas Health Sci. Ctr., Houston, TX 77030), and Christy K. Holland (Univ. of Cincinnati, Cincinnati, OH 45267-0586)

Echogenic liposomes (ELIP) have been developed for drug encapsulation. The gas contents in ELIP present a potential mechanism for ultrasound-triggered release of drug contents. Calcein, a fluorescent dye, was loaded in ELIP as a drug substitute (C-ELIP) and ultrasound-induced release was quantified with fluorescence spectrophotometry. Pulsed 6.0-MHz color Doppler from a clinical diagnostic ultrasound scanner (CL15-7 transducer, Philips HDI 5000, MI of 1.3, 150 Hz pulse repetition frequency) was applied to samples of C-ELIP in an *in vitro* flow phantom (2.2 ml/min). For comparison, Triton X-100 was added to C-ELIP to release calcein. Control samples of C-ELIP were not treated with ultrasound or Triton X-100. The echogenicity of C-ELIP (expressed as mean digital intensity in a 0.5 cm^2 region of interest) decreased by 9.6 ± 2.1 dB after exposure to ultrasound. The observed calcein concentration (μg/ml) was 3.1 ± 0.1 for the untreated sample, 4.5 ± 0.1 after Triton X-100 treatment, and 4.2 ± 0.2 after ultrasound exposure. 65.6% of encapsulated calcein was released with ultrasound. These results demonstrate that ultrasound-mediated release of drugs from ELIP using a clinical diagnostic ultrasound scanner is feasible. [Work supported by an AIUM Education and Research Grant and NIH 1R01 HL074002, and NIH 1RO1 NS047603-01S1.]

9:15

3aBB5. Acoustic targeted drug delivery in neurological tissue. George Lewis, Jr., William Olbricht (Cornell Univ., 108 Olin Hall, Ithaca NY 14850, george@cornellbme.com), and George Lewis, Sr. (Transducer Eng. Inc, P.O. Box 4034, Andover MA 01810, thearrayman@transducerengineering.com)

The success of treating brain cancer such as neuroblastomas and neurofibromatosis has not been very effective, and is in fact the leading cause of cancer-related death in patients younger than age 35. In the last 10 years recent developments in drug delivery methods have allowed doctors to implant/inject time-release drugs into the tumor cavity that allows for continuous release of chemotherapy; however results from these studies have not been as successful as anticipated. It is believed that the non-treated cancerous cells are able to migrate from the original tumor site, and relocate beyond the diffusion range for effective drug treatment. In this study we utilize high frequency focused ultrasound to increase drug perfusion into phantoms that mimic brain tissue as a method to increase the rate of drug permeation into the tissue before vascular clearance and cell migration reduce its effectiveness. Using various acoustic pulse sequences we show a substantial increase in drug perfusion into brain mimicking tissue then can be achieved by diffusion alone. We are therefore able to reduce the time of delivery, the amount of drug delivered and the drugs local impact range. This could significantly increase the success of treatment and reduce systemic effects of chemotherapy.

9:30

3aBB6. Robotic high intensity focused ultrasound (HIFU) control for tumor treatment. Shivkumar Kambhampati and Vesna Zderic (The George Washington Univ., 801 22nd St. NW, 624E, Washington, DC 20052, shivak@gwu.edu)

Introduction: Robotic surgery, with more precise and defined movements compared to manual human surgery, is the wave of the future. The purpose of this project is to move the focus of the HIFU transducer using a 3-axis robotic work cell in order to treat the whole volume of a tumor in a precise manner. Methods: A 3-axis robotic work cell from Arricks Robotics was chosen based on price ($3300 USD), precision, range of movement, and payload capacity (up to 2Kg). A LabVIEW program serves as the entire surgery interface including movement options, treatment algorithms with up to 25 points within the tumor, and a guidance system via ultrasound imaging and the hyperechoic spot of the HIFU focus. Results: The robotic work cell could achieve 0.5mm per 10mm precision and speeds of 10mm/sec in real-time within a cell of 180mm by 180mm by 50mm with a HIFU transducer mounted. Testing with native MD-2xp software has resulted in distance testing with less than 2% error in all three axes (0.7% in X, 1.5% in Y, 1.3% in Z). Conclusion: The final outcome will provide a robust, computer-assisted, cost-effective way to perform robotic HIFU surgery.

9:45

3aBB7. Comparison of pathway in high-intensity focused ultrasound lesion production. Yufeng Zhou, Joo Ha Hwang (Div. of Gastroenterology, School of Medicine, Univ. of Washington, Seattle, WA 98195), Kwang Kim, and Joo Ha Hwang (Univ. of Washington, Seattle, WA 981095)

High intensity focused ultrasound (HIFU) is being evaluated for non-invasive treatment of solid tumors. The temperature at the HIFU focus can reach over 65 °C, denaturing cellular proteins and resulting in coagulative necrosis and lesion formation. One common method for delivering HIFU therapy clinically is using the spot accumulation method that delivers sequential individual treatment spots. Because of thermal diffusion from nearby treatment spots, the size of subsequent lesions will gradually become larger as the HIFU therapy progresses, which may cause insufficient treatment of the initial spots, and over-treatment of later spots unless parameters are changed during treatment. A new pathway for HIFU treatment is proposed and compared with the conventional sequential path. Modeling, in vitro phantom and ex vivo bovine liver experiments demonstrate that the new treatment path produces more uniform lesions than the

3a THU. AM

conventional treatment path ($p < 0.05$). The relationship between lesion area/volume and delivered ultrasound energy and the dose-dependent discrepancies between scanning paths were also studied. In addition, the temperature changes in the ex vivo system were measured using a thermocouple array. Altogether, the new treatment path appears to be advantageous for producing more uniform lesions without modifying HIFU parameters during treatment or significantly increasing the scanning time.

10:00–10:15 Break

10:15

3aBB8. Optimization of lesion formation using high-intensity focused ultrasound at large tissue depths. Joshua Samuels and Vesna Zderic (Dept. of Elec. and Comput. Eng., George Washington Univ., 725 23rd St NW, Washington, DC 20052, joshasam@gwu.edu)

Background: The objective was to determine the parameters to making a sizable, controlled lesion deep within the treated tissue using high-intensity focused ultrasound (HIFU). High HIFU intensities allow for rapid temperature rises in tissue used for tumor ablation. Methods: Various experiments were conducted manipulating amplitude and treatment time to produce lesions at full focal length (5.2 cm) of a 3.35 MHz HIFU transducer. Beef thigh was used, as it closely resembles the nonuniformity of human tissue unlike phantom-gels or turkey breasts. Results/Discussion: Inconsistent lesion formation in our experiments showed that deep lesions in a nonuniform medium are not easily created. At 5.2 cm, in situ HIFU intensities dropped to 170–260 W/cm^2 (vs. 40,000–60,000 W/cm^2 free field). Reflections off fat, fascia, and bubbles formed at the HIFU focus (at higher intensities) often appear to result in prefocal lesion formation (average total depth of 3.8 cm). Lower intensities over longer treatment times (up to 120 s) yielded desirable lesion depth (average depth of 4.5 cm, highest depth of 6.8 cm), showing longer treatments at lower intensity could be the key to precise deep lesions. Lesion volumes ranged from 0.1 to 26.5 cc. Conclusion: The challenges posed by treatment of nonuniform tissues are similar to those to be encountered in clinical applications.

10:30

3aBB9. Enhanced high-intensity focused ultrasound lesion formation in tissue phantoms using embedded fibers. Cecille Labuda and Charles C. Church (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, cpembert@olemiss.edu)

High intensity focused ultrasound (HIFU) is currently being developed for hemorrhage control, since it provides rapid energy deposition in the focal region of the field. Near large vessels, the deposition rate is limited by loss of heat to the blood flow, making hemorrhage control difficult. In this study, nylon, copper, and stainless steel fibers were embedded in an albumen-containing tissue phantom to investigate whether these fibers would enhance the HIFU heating effect. The embedded fibers were sonicated at high power, and control sonications were performed away from the fibers. Visible regions of protein denaturation, or lesions, were produced. Control lesions were ellipsoidal and elongated along the acoustic axis. Lesions at the stainless steel and copper fibers were similar in shape, but smaller than the control lesions. Lesions produced at the nylon fibers were teardrop-shaped, elongated along the fiber, and larger than the controls. The roles of thermal conductivity and ultrasonic absorption in the enhanced heating effect were considered. It was concluded that a material of low thermal conductivity and high ultrasonic absorption, such as a polymer fiber, was most suitable for enhancing the HIFU heating effect. [Work supported by DAMD17-02-2-0014.]

10:45

3aBB10. Cost-effective radiation force balance for calibration of therapeutic ultrasound devices. Faezeh Razjouyan and Vesna Zderic (The George Washington Univ., 6882 Lafayette Park Dr., Annandale, VA 22003, faezeh@gwu.edu)

The objective was to create an inexpensive, portable, and accurate absorptive radiation force balance to measure acoustic powers of up to 100 watts generated by the high intensity focused ultrasound (HIFU) transducer. This paper describes the process of making an effective absorbing target with commercially available ingredients. Four different absorbing targets consisting of nickel powder, silicone elastomer, and microballoons were prepared and tested. Silicone Sylgard (Dow Corning, MI) was used for all samples. However, two different microballoons (acrylic and phenolic) and two different nickel powders (high density and spherical nickel powder) were used. The final results were compared with a commercially available reflection radiation force balance (RRFB). The results for the same 3.5 MHz HIFU transducer (Sonic Concepts, WA) revealed that a combination of spherical nickel powder (Alfa Aesar, MA) with acrylic microballoons (Douglas Sturgess, CA) offered an average efficiency of 89.8%, compared to that of RRFB, which was 80.7%. A combination of high density nickel powder (Inco Inc., Canada) with acrylic and phenolic microballoons were 82.5% and 84.2%, respectively (for the same HIFU transducer), while spherical nickel powder and phenolic microballoons (US Composites, FL) had efficiency of 64.4%, indicating incorrect measurements of HIFU transducer efficiency.

11:00

3aBB11. Characterization of low-profile fresnel lenses for annular high-intensity focused ultrasound radiators. David Woolworth, Jason Raymond, and Joel Mobley (Natl. Ctr. for Physical Acoust., Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677, jmobley@olemiss.edu)

As part of an effort to develop high intensity focused ultrasound (HIFU) applicators for remote hemostasis, we are investigating the use of low-profile stepwise Fresnel lenses for varying the focal depths of single-element annular HIFU transducers. The Fresnel lenses are interchangeable and represent a cost-effective approach to variable HIFU focusing in comparison to multi-element phased array systems. In this work, we report on the characterization of a Fresnel lens coupled with a 1.2 MHz annular transducer as the inner radius of the annulus is varied. We present data derived from hydrophone-based measurements of the pressure field of the transducer/lens system. In addition, we compare our findings with simulations derived using the angular spectrum technique.

11:15

3aBB12. Detection of brachytherapy seeds at varying angles with a singular-spectrum-analysis algorithm. Sarayu Ramachandran, Jonathan Mamou, and Ernest J. Feleppa (F. L. Lizzi Ctr. for Biomed. Eng., Riverside Res. Inst., 156 William St., New York, NY 10038)

A singular-spectrum-analysis (SSA) algorithm previously was shown to be successful in detecting brachytherapy seeds in B-mode images when seeds were orthogonal to the ultrasound beam. In this study, the algorithm was extended to detect seeds at angles to the beam. The SSA algorithm derives P-values from selected eigenvalue pairs of the autocorrelation matrix of seed echo signals. The P-value indicates the probability of presence of a seed. A seed inserted in a gel pad and another inserted in beef tissue were scanned with a 5-MHz transducer. The seed angle was varied from orthogonal to the beam to 23 deg from orthogonal, using 1-degree increments. Simulations of seed echoes based on empirical data were generated and used to test the algorithm. Success in seed detection was expressed by a score computed from P-values. Scores for simulations decreased from 0.8 to 0.45 as the angle increased from 0 to 12 deg. Scores for experimental data varied from 0.74 at 0 deg to 0.62 at 23 deg; the lowest value was 0.1 at 7 deg. Accordingly, the SSA algorithm was successful in detecting the seed with clinically relevant angles between seed and beam.

11:30

3aBB13. Flexible tools for time reversal acoustics focusing applications. Laurent Fillinger (Stevens Inst. of Technol., Castle Point on Hudson, Hoboken, NJ 07030-5991, laurent.fillinger@artannlabs.com), Viktors Kurtenoks, Sam Rosenblum, Alexander Sutin, and Armen Sarvazyan (Artann Labs., West Trenton, NJ 08618)

Time reversal focusing of acoustic waves has numerous biomedical applications including medical imaging, HIFU treatment, and targeted drug delivery. These various applications pose different requirements. Imaging requires a small spatial focused spot, which can be scanned in 2-D or 3-D; therapy requires high deposited energy and controlled-shape extended focus. Artann Laboratories is developing flexible tools including hardware, algorithms, and software with user-friendly interface as a universal platform for wide-range applications, based on the use of the time reversal focusing of acoustic waves. We have built a ten-channel electronic system with sampling frequency up to 33.3 MHz. It enables simultaneous broadcasting of independent arbitrary signals and recording of the response by a scanning receiver/beacon. A set of functions were developed to enable the control of this setup within Matlab, which offers a high level computing language for numerical computation. With a developed system, a diverse variety of experiments and data analysis can be conveniently conducted within the same platform. Illustrative results on numerous biomedical applications of the time reversal focusing obtained within this framework will be presented.

THURSDAY MORNING, 29 NOVEMBER 2007 NAPOLEON B1, 10:30 A.M. TO 12:00 NOON

Session 3aEA

Engineering Acoustics and Underwater Acoustics: Underwater Transduction

Elizabeth A. McLaughlin, Chair
Naval Undersea Warfare Center, 1176 Howell St., Newport, RI 02841

Contributed Papers

3a THU. AM

10:30

3aEA1. An investigation of the capabilities of a short hydrophone array towed by an ocean glider. Georges A. Dossot, James H. Miller, Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett Bay Campus, Narragansett, RI 02882, georges@oce.uri.edu), Edmund J. Sullivan (EJS Consultants, Portsmouth, RI), Jason D. Holmes (BBN Technologies, Cambridge, MA), James F. Lynch (Woods Hole Oceanograph. Inst., Woods Hole, MA), and Scott Glenn (Rutgers Univ., New Brunswick, NJ)

During the Shallow Water Experiment (SW06) a Webb Slocum glider, deployed by Rutgers University, demonstrated that gliders are promising vehicles for towing short acoustic arrays. The gliders saw-tooth trajectory allows for sampling the water column at varying depths and ranges. Further, the glider provides a low-speed platform, allowing for a flow-noise free towed array, which is ideal for the processing of low level signals. One attractive application for glider-towed arrays is target tracking. By using the passive synthetic aperture effect, coupled with a near-field model for the signal, the coordinates of an acoustic source can be estimated using a Kalman filter, but without the necessity of the maneuver normally required by bearings-only tracking. This is possible since the large aperture traced out by the glider permits wavefront curvature to be exploited for range estimation. Using synthetic narrowband data, it is shown that the range and bearing of a low-level acoustic source can be estimated without changing the gliders course. The algorithm is based on an Unscented Kalman Filter. Also, an approach for the broadband problem is outlined. [Work sponsored by the Office of Naval Research.]

10:45

3aEA2. Impact of structural support on vector sensor acoustic performance. David Deveau (PSC 1012, Box 701, FPO, AA 34058, david.deveau@autec.navy.mil)

Vector sensors hold great promise for the exploitation of undersea acoustics by providing the ability to measure the acoustic intensity. Current sensor developments can measure the acoustic acceleration in three dimensions as well as the pressure, which when combined through a cross-correlation produces the acoustic intensity field surrounding the sensor. Mathematical studies indicate that a gain of 4.6 dB is achievable by beamforming the sensor's three cardioid outputs with the pressure sensor's omnidirectional response. In these studies, the mechanics of physically retaining the sensor in position is not considered, but is necessary to allow the sensor to freely sense particle velocity/acceleration and not shadow the sensor to the surrounding pressure field. A set of PVC frames was constructed to hold an individual sensor, which was then combined to form an array of sensors for use in a long-term shallow water acoustic field intensity study. These frames are modeled to determine the level of forward and backscatter caused by the flooded PVC pipes and the possible impact to the theoretical patterns. The physical structures are then calibrated in a controlled measurement facility to determine if the accelerometers move freely in all three dimensions to generate the cardioid patterns necessary to achieve the gains predicted by intensity models over similar pressure sensor configurations.

11:00

3aEA3. Coverage metric for track planning and location of stationary bottom targets. Steven M. Dennis (Code 7183, Naval Res. Lab., Stennis Space Ctr., MS 39529, steven.dennis@nrlssc.navy.mil)

With the increasing dependence on optimization algorithms in track generation in the presence of variable or uncertain environments, the desire for a more environmentally-sensitive metric of sensor performance has arisen. A measure of performance based upon acoustic coverage area has shown promise as a basis for sensor utilization and optimized track generation applications, and is presented here for the case of a bi-static, active sensor used in the search for stationary bottom targets (e.g., wrecks). Coverage can be defined as the areas throughout which a sensor has a sufficiently high signal-to-noise ratio or alternatively, probability, of making positive observations of objects on the seafloor. For the purposes of determining optimal sensor placement and track generation, the area of interest is divided into a sufficiently sampled grid of calculation points. Computing and compiling acoustic sensor coverage area information for grid points throughout the area of interest into an acoustic coverage map gives immediate visual feedback on locations of optimal performance for a specific sensor in the current ocean environment. [The authors appreciate and acknowledge the funding support from the Naval Research Laboratory Base Program.]

11:15

3aEA4. Enhancing underwater acoustic vector sensor measurement performance by point source analysis. Joseph A. Clark (NSWCCD, Code 7340, 9500 MacArthur Blvd., West Bethesda, MD 20817-5000)

The beam pattern response to an actual point-like source that is measured with a vector sensor has been found to differ noticeably from the computed response to an idealized point source even under very low ambient noise conditions. This difference can be exploited to enhance underwater acoustic vector sensor measurements by a repeated process of subtracting an equivalent point source response from the measured response and analyzing the remainder beam. Details of a method for carrying out this type of analysis will be presented. The method will be illustrated with data from an at-sea experiment conducted at a U. S. Navy radiated noise facility near Ketchikan, Alaska (SEAFAC). Features of the equivalent point sources determined by the method will be shown to correspond to known characteristics of the ambient noise field at the measurement site. An evaluation of the improvement in detection and localization performance which can be achieved by the point source analysis method will be reported.

11:30

3aEA5. Acrylic plate acoustic transmission experiment and theory. David R. Dowling and Natasha A. Chang (Dept. of Mech. Eng., Univ. of Mich., 2350 Hayward St., 2212 G. G. Brown Lab., Ann Arbor, MI 48109)

An approximate model of the transmission of sound waves in water through a plate of PMMA was developed for a point source. The model is based on spherical wave propagation and plane-wave transmission through a solid layer, and it is calibrated by identifying the properties of the plastic. By minimizing the error between the modeled and the experimentally measured acoustic wave, the mechanical properties of PMMA can be estimated for PMMA plates of varying thickness d for the d/fluid range of 0.04 to 2.5, and sound-incident angles of 0 deg to 35 deg. This calibration requires only the compression wave ray-path information and was shown to achieve 90% correlation between experimental and predicted waveforms for synthetic cavitation pulses with a nominal bandwidth from 40 kHz to 200 kHz. At larger angles of incidence as measured based on the compression wave, it was necessary to track the various waves that occur in the solid, i.e., compression, shear, and evanescent. The acoustic pressure waves that they generate have to be appropriately added at the receiver location to then recreate the experimentally-measured acoustic signal. The achieved model-experiment correlation at these greater angles of incidence was 80%. [Work supported by ONR.]

11:45

3aEA6. Development of a flextensional transducer for high resolution seismic imaging. Bertrand Dubus, Gerard Haw, Christian Granger, Pascal Mosbah (IEMN dpt ISEN, UMR CNRS 8520, Lille, France), Arkadiusz Kosecki, Bogdan Piwakowski (IEMN, Villeneuve, d'Ascq, France), and Patrick Meynier (Institut Francais du Petrole, Rueil Malmaison, France)

Piezoelectric transducers are considered as potential sources for geophysical applications such as 4D monitoring or shallow subsurface survey, which require high repeatability and precise control of the emitted signal. However, the use of piezoelectric sources is limited by their low power and narrow bandwidth. Flextensional transducers are compact, high power, low frequency, wide-band projectors used in underwater acoustics. The design, fabrication and test of a flextensional transducer for high resolution seismic imaging are presented in this paper. Circuit modeling is used to discuss the general properties of piezoelectric sources radiating in the ground. Finite element simulations (ATILA code) are conducted to design a wide band 500–1500 Hz flextensional transducer. On-site tests are performed for a typical configuration of shallow subsurface survey. The prototype transducer is compared to a weight drop which constitutes a well-known seismic source. By taking advantage of the source repeatability and using specific signal processing, it is found that the flextensional transducer is equivalent to weight drop ranging from 300 to 1300 J. Bandwidth is increased by 200 to 300%, leading to a significant improvement of image resolution.

THURSDAY MORNING, 29 NOVEMBER 2007 NAPOLEON A3, 8:00 TO 11:30 A.M.

Session 3aNS

Noise: Soundscape Developments: Case Studies and Best Practices

Brigitte Schulte-Fortkamp, Cochair
Technical Univ. Berlin, Inst. of Fluid Mechanics and Engineering, Secr TA 7, 10587 Berlin, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 27 Hartford Turnpike, Vernon, CT 06066

Chair's Introduction—8:00

Invited Papers

8:05

3aNS1. Better soundscapes for all—Report on the Workshop in Standardization for Soundscape Techniques held in Salt Lake City, 5 June 2007. Brigitte Schulte-Fortkamp (TU-Berlin ISTA, Einsteinufer 25, D-10587 Berlin, Germany, brigitte.schulte-fortkamp@tu-berlin.de) and Bennett M. Brooks (Brooks Acoust. Corp., Vernon, CT 06066)

The goal of the Soundscape Workshop in Salt Lake City (153rd ASA meeting) was to explore standardization as a possible future advancement in the evolving field of Soundscape measurement, analysis, and design. The perception of the soundscape can provide comfort, tranquility, and needed information to the person concerned, or may be a source of annoyance. The combination of physical acoustical measurements with scientific evaluation of perceptual responses to environmental sound, known as soundscaping, is an essential method for the assessment and actualization of positive outdoor environments. Engineering and aesthetic soundscape design may only proceed based on a complete characterization of the acoustical environment, including the nature of the sound sources and the reactions of the perceivers. Soundscaping provides for the measurement, analysis, and design of environmental sound by applying the knowledge of both science and community experts. Much fundamental and practical research has been conducted to establish the bases for the soundscape field. The next step is for researchers and practitioners to standardize the available soundscape techniques to allow for more comparison of test and survey results and wider application in design. This presentation outlines the proposed methods and means developed at the workshop for follow-up and further action.

8:30

3aNS2. Soundscapes without decibels. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601)

This paper will examine the advantages and limitations of describing soundscapes in terms of the effects of on specific populations.

8:55

3aNS3. Identification of distinctive patterns and features in soundscapes. Klaus Genuit and Andre Fiebig (HEAD acoustics GmbH, Eberstrasse 30a, 52134 Herzogenrath, Germany, klaus.genuit@head-acoustics.de)

The environmental noise of several cities located all over the world was recorded, compared, and analyzed. The analyses were focused on the determination of distinctive features and significant patterns of the respective soundscapes, which led to the unique sound of the urban places and that are important because of their individuality, numerousness, or domination. By means of this study and the applied analyses, several acoustical properties, particularities, and patterns were identified, which could be the starting point for an acoustical-orientated soundscape classification. It was found that the environmental noise differs, for example, in character, spectral content, time structure, degree of variation of certain parameters, background to foreground relationship caused by the location-specific noise sources, and their spatial as well as temporal occurrence. It is important to emphasize that the acoustical analyses, on the one hand, have to capture the global, overall impression of the soundscape and, on the other hand, must recognize and adequately interpret single noise events, which also cause strong reactions and emotions. The presentation will show and discuss the case study results.

9:20

3aNS4. From global and semantic evaluations to physical measurements and modeling. Daniele S. Dubois (LCPE/LAM/IJLRDA, 11 rue de Lourmel, 75015 Paris, France, ddubois@ccr.jussieu.fr)

Soundscape research has now largely pointed out the limits of indicators solely (directly) grounded in pressure level measurements and other simple static acoustic parameters to account for nuisance as well as sound quality, as perceived by human listeners. Advances in cognitive research has allowed to identify semantic categories of soundscapes contrasting pleasant areas versus annoying ones with source identification and evaluation as one major criterion of categorization. The present challenge, both theoretical and pragmatic, is therefore to connect such global semantic (psychological) evaluations to analytical (physical) instrumental measurements. We suggest that physical modeling of source signatures and artificial intelligence models of source identification processes could now provide formal representations of typical environmental categories, that could be the missing link to fill the gap between these two types of sound descriptions and interpretations. We will present a state of art concerning our present day knowledge of the structural properties of categorical structures we identified in psychological and semantic investigations, to specify the conditions under which cognitive modeling could account for the structural complexity of source contribution to soundscapes evaluation.

9:45

3aNS5. "The Grand Canyon" vs. "Soundscape of Nowhere." Dickson J Hingson (SIERRA CLUB—Natl. Parks and Monuments Committee, 275 River Run Rd., #3, Flagstaff, AZ 86001)

Twenty years have elapsed since the National Parks Overflights Act of 1987 targeted the once quiet but aviation-imperiled Grand Canyon National Park soundscape as worthy of an expeditious, potentially difficult "substantial restoration." Within several more months, this protracted acoustic effort reaches its long-standing presidential deadline (April, 2008) for completion, as according to specifications/standards set by the National Park Service. A preliminary assessment (first presented by this author at ASA's June biennial meeting) will be refined/updated in view of subsequent developments. Primary and emerging supplemental noise indicators and long-established (or possibly revised) Park Service restoration standards-based mainly on audibility and "restored" acreage—will be reviewed. Pertinent words and dimensions of soundscape assessment from this and other soundscape studies will be reviewed and compared, particularly re concepts of "wilderness character" and "beauty." Effectiveness of anticipated management actions in the face of ongoing political/legal controversy will be examined, pitting restoration of the authentic Grand Canyon wilderness soundscape against a relatively unsavory option, "The Soundscape of Nowhere." The Grand Canyon soundscape situation situation will be compared with preservation/restoration needs facing other similarly imperiled, iconic national parks in the West, which are at continuing risk for long-term aviation noise impairment.

10:10–10:25 Break

Contributed Papers

10:25

3aNS6. Development of soundscape assessment and design methods for large institutional projects. Gary W. Siebein, Robert M. Lilkendey, and Hyun Paek (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607)

Soundscape assessment and design processes were used to develop design criteria for noise mitigation strategies for the expansion of an institutional complex in a medium-sized city. A series of long-term acoustical measurements of average A-weighted sound levels in the community as well as more detailed octave band measurements and calibrated aural recordings of specific acoustic events that comprised the ambient sounds were mapped for the neighborhood. Sound walks were conducted at various times of day to understand the dynamics of the acoustical environment and to identify issues. Focus group discussions among stakeholders and team members developed the long-term plans for the community and appropriate architectural and acoustical design criteria for the project. Analysis of auralizations of various design options by stakeholders and team members using sophisticated multi-channel playback systems and full-scale aural demonstrations on site were used as part of the evaluation process.

10:40

3aNS7. Toward a less chaotic sound environment for nurses: A study investigating the relationship between layout design, aural connectivity, and user activities. Sele Okcu and Craig Zimring (College of Architecture, Georgia Inst. of Technol., Atlanta, GA 30332-0155)

An intensive care unit has a particularly demanding aural environment, where nurses must be able to hear and respond to numerous alarms, hear critical orders, and respond to patients and family members. This study explores a network measure called aural connectivity that reflects the overall pattern of where users can hear and respond to all key sounds. Aural connectivity is explored in a 20-bed neuro-ICU in a major teaching hospital by mapping locations where each alarm and key sound from the patient rooms can be heard, using both self-report and objective measures of sound. This study explores how aural connectivity relates to tasks, unit layout and design, and the nature of noise sources. Aural connectivity has the promise of being a useful overall design and evaluation tool because it allows movement and tasks to be mapped onto a network of where effective auditory monitoring can occur.

10:55–11:30
Panel Discussion

THURSDAY MORNING, 29 NOVEMBER 2007 BORGNE, 8:25 TO 10:00 A.M.

Session 3aPAa

Physical Acoustics: Ultrasound Measurements and Methods for Condensed Matter

Veerle M. Keppens, Chair
Univ. of Tennessee, Materials Science and Engineering, Knoxville, TN 37996

Chair's Introduction—8:25

Invited Papers

8:30

3aPAa1. Slow dynamics and the Larsen effect at millisecond time scales. Richard Weaver and Oleg Lobkis (Dept. of Phys., Univ. of Illinois, 1110 W Green St., Urbana, IL 61801)

At sufficient gain, an ultrasonic feedback circuit rings with a ''Larsen'' tone that depends on the nonlinear electronics, and also on the acoustic properties of the solid body to which it is attached. Because the spectrum of this tone is extraordinarily narrow and stable, it may be measured with high precision. With a goal of using this to monitor small rapid changes in materials, here the stability of the signal to perturbations is quantified, and the results applied in the monitoring of the evolution of the effective modulus of mesoscopically elastic bodies (cements and stones) after brief transient loads. In accord with other studies, it is found that the modulus drops after the load, but then recovers in a characteristic manner, like $\log(t)$. The present technique, using as it does frequencies of order MHz and loads with durations of order 10 μs, extends these studies of slow dynamics to early times. It is found that $\log(t)$ behavior can be sustained over the full range investigated, from a couple of μs to hundreds of seconds. [Work supported by NSF CMS 05-28096.]

8:55

3aPAa2. Elastic constant measurements in heterogeneous systems: Thin films on a substrate. Joseph Gladden III (Dept. of Phys. and Astron., Univ. of Mississippi, 108 Lewis Hall, University, MS 38677, jgladden@phy.olemiss.edu)

Resonant ultrasound spectroscopy (RUS) has shown itself to be an efficient and accurate method for determining the full elastic tensor of a single crystal sample. Elastic constants are a measure of the curvature of the interatomic potentials, and are thus sensitive to a variety of phase transitions in materials. In this talk, an extension of RUS will be presented that allows for the determination of the elastic constants of a crystalline thin film deposited on a substrate with thicknesses on the order of 100 nm or greater. Both

experimental and numerical analysis issues for thin film measurements will be addressed. Some "proof of principle" results will be presented as well as experimental data showing magnetic phase transitions in colossal magnetoresistance films between 200 nm and 400 nm thick. Thin film RUS should prove to be a useful tool in the effort to better understand phase transitions in reduced dimensionality and effects of lattice mismatch between substrate and film.

9:20

3aPAa3. Making resonant ultrasound spectroscopy measurements from milliKelvin temperatures to well above room temperature. J. B. Betts, A. Migliori, and I. Stroe (NHMFL, Los Alamos Natl. Lab. Los Alamos, NM 87545, jbbetts@lanl.gov)

Measuring elastic moduli using resonant ultrasound spectroscopy (RUS) can be very challenging at the best of times. When the measurement environment needs to be controlled from milliKelvin to many hundreds of Kelvin, this challenge becomes extremely acute. We will present methods and results from such experiments showing that with care, elastic moduli can be very accurately determined over these extreme temperature ranges. [Work supported by the National Nuclear Security Administration, U.S. Dept. of Energy, State of Florida, and the National Science Foundation.]

Contributed Paper

9:45

3aPAa4. Common acoustic properties of solid aggregate at micro- and nano-scales. Hasson M. Tavossi (Dept. of Phys., Astron., and Geosciences, Valdosta State Univ., 1500 N. Patterson St., Valdosta, GA 31698)

Acoustic properties of solid aggregate, when measured at macroscopic scale, have been found experimentally to have remarkable similarities with the same wave properties observed at the atomic and nano-scales. It can be shown that the elastic moduli and other wave properties such as, wave tunneling, attenuation, cutoff-frequency, and dispersion, depend on the same structural factors at macro- and nano-scales. The model constructed for the acoustic properties of solid aggregate, expressed in terms of wave-number (ka), can be applicable in the wide range of frequencies and length scales. The findings on acoustic properties of solid aggregate at macroscopic scale could lead to a better understanding of the wave properties of solids at nanoscales. These readily analyzable models for acoustic properties at macroscopic scales are convenient tools for verification of theoretical models for acoustic behavior of complex solids at atomic level. Experimental data and numerical results are compared, showing acoustic wave responses of the solid aggregate for a wide range of frequencies, from audible to ultrasound. The relevance of these results to nano systems will be discussed to show that parallel acoustic properties can exist in the range of characteristic lengths from macro- to nano-scales.

THURSDAY MORNING, 29 NOVEMBER 2007 BORGNE, 10:15 A.M. TO 12:00 NOON

Session 3aPAb

Physical Acoustics: Thermoacoustics

William V. Slaton, Chair

Univ. of Central Arkansas, Dept. of Physics and Astronomy, 201 Donaghey Ave., Conway, AR 72035-0001

Contributed Papers

10:15

3aPAb1. Shape factor characterization of fibrous media with a temperature gradient. William V. Slaton (Dept. of Phys. & Astron., The Univ. of Central Arkansas, Conway, AR 72035, wvslaton@uca.edu)

Recent theoretical work generalizes thermoacoustic theory to random porous media [H. S. Roh et. al., J. Acoust. Soc. Am. **121**(3), 1413–1422 (2007)]. Characteristics of the porous media, such as the tortuosity and dynamic shape factors for viscous and thermal effects, are introduced into the thermoacoustic wave equation and may be determined by suitable impedance measurements at zero temperature gradient. This theoretical approach may also be used to model fibrous media, such as fiberglass or steel wool. A new technique to determine the scaling factors for fibrous media utilizing a single-step finite difference inversion of the thermoacoustic wave equation [R. Raspet et. al., J. Acoust. Soc. Am. **103**(5), 2395–2402 (1998)] with zero temperature gradient will be presented. Roh et. al. predict that acoustic gain with a nonzero temperature gradient may be written in terms of these scaled cylindrical dissipation functions. The acoustic gain term may also be determined by suitable application of the single-step finite difference technique. [This material is based upon work supported by the U. S. Army Space and Missile Defense Command under Contract No. W9113M-06-C-0029.]

10:30

3aPAb2. Simulating thermoacoustics in random stack materials. Carl Jensen (Dept. of Phys. and Astron. and the Natl. Ctr. for Physical Acoust., Univ. of Mississippi, University, MS 38677)

A lattice Boltzmann based computational fluid solver is being developed to investigate the thermoacoustic properties of stack materials with irregular pore geometries. The goal of the investigation is to perform a direct simulation of the time-varying thermohydrodynamic flow within a porous material and determine the material's thermoacoustic properties from its base characteristics. Accurately simulating a porous material's pore scale flow requires methods for handling the irregular boundary conditions presented by the material and the use of high performance computing to handle the large number of elements required to represent the material's bulk behavior. Progress in both of these areas will be presented as results in applying the use of commodity graphics programming units for high performance computing and a local grid refinement scheme for increased resolution at the wall boundaries. [Work supported by U.S. Army Space & Missile Defense Command.]

10:45

3aPAb3. Design environment for low-amplitude thermoacoustic energy conversion (DeltaEC). John P. Clark, William C. Ward, and Gregory W. Swift (M.S. K764, Los Alamos Natl. Lab., Los Alamos, NM 87545, swift@lanl.gov)

The Los Alamos thermoacoustics code, available at www.lanl.gov/thermoacoustics/, has undergone extensive revision this year. New calculation features have been added to the original Fortran computational core, and a Python-based graphical user interface wrapped around that core provides improved usability. A plotter routinely displays thermoacoustic wave properties as a function of x or tracks results when a user-specified input variable, such as frequency or amplitude, is varied. The Windows-like user interface provides mouse-based control, scrolling, and simultaneous displays of plots and of several categories of numerical values, in which color indicates important features. Thermoacoustic phenomena can be calculated with superimposed steady flow, and time-averaged pressure gradients are calculated. In thermoacoustic systems with toroidal topology, this allows modeling of steady flow caused by gas diodes (with or without time-averaged heat transfer) and Gedeon streaming. Thermoacoustic mixture separation is included, also with superimposed steady flow. The volume integral of the complex gas momentum is available, so vibrations of thermoacoustic systems can be analyzed.

11:00

3aPAb4. Thermoacoustic quality factor measurement in a Helmholtz resonator. Holly Smith and William Slaton (Dept. of Phys. & Astron., The Univ. of Central Arkansas, Conway, AR 72035)

A Helmholtz resonator consists of a hollow neck attached to an empty chamber. This resonator can be modeled as a spring-mass system in which the air moving inside the neck acts as the mass and the gas inside the chamber acts as the spring. Every Helmholtz resonator has a characteristic quality factor that is dependent upon the total mechanical resistance present. A system with low resistance will have a narrow peak on its amplitude versus frequency graph and a high quality factor, whereas a system with high resistance will have a broader peak and a low quality factor. In this experiment, a porous ceramic substrate is inserted into the neck of a Helmholtz resonator. Introducing this substrate into the neck of the Helmholtz resonator alters the resonance frequency and the quality factor of the resonator. The quality factor will be shown to be increased with increased temperature difference until spontaneous onset of sound generation. The effect of convection will be studied by exploring the dependence of the quality factor on the physical orientation of the resonator with respect to the vertical. [This work was supported by the Arkansas Space Grant Consortium and the University of Central Arkansas University Research Council.]

11:15

3aPAb5. Effect of neck geometry on aeroacoustic excitation of a Helmholtz resonator. Stephanie Lanier and William Slaton (Dept. of Phys. & Astron., The Univ. of Central Arkansas, Conway, AR 72035)

The aeroacoustic response of a Helmholtz resonator, when attached to a wind tunnel via junctions with different neck geometries, has been examined. The wind tunnel consists of a 2-inch ID glass tube with variable mean flow at atmospheric pressure connected to a cross-junction with two coaxial 5-liter Helmholtz resonators. During the experiment, the acoustic pressure and corresponding frequency is measured in one of the Helmholtz resonators at different wind tunnel flow velocities. By knowing these quantities, the acoustic velocity of the air at the neck of the resonator can be determined. The ratio of the acoustic velocity to the flow velocity as a function of the Strouhal number can be studied for differing neck geometries. Helmholtz resonator neck geometries studied include 90-deg bends and straight sections in various combinations. [This work was supported by the Arkansas Space Grant Consortium and the University of Central Arkansas University Research Council.]

11:30

3aPAb6. An analytical model for the design of optimized heat exchangers for thermoacoustic systems. Arganthaël Berson, Philippe Blanc-Benon (LMFA, UMR CNRS 5509, Ecole Centrale de Lyon, 36 avenue Guy de Collongue, 69134 Ecully Cedex, France), and Vitalyi Gusev (Université du Maine, 72085 Le Mans Cedex 9, France)

A 1-D nonlinear model for the heat transport from the edge of a thermoacoustic stack to a heat exchanger is developed. This model is an extension of the previous study by [Gusev et al., J. of Sound and Vibration **235**, (2000)] to the case of a finite dimension heat exchanger with a temperature difference between the stack and the heat exchanger plates. The model is based on a relaxation-time approximation for transverse heat transfer that allows for the calculation of the heat flux through an adiabatic gap, separating the stack and the heat exchangers. It shows the generation of temperature harmonics close to the edge of the plates that leads to nonlinear heat transport through the gap. The heat flux extracted by the hot side heat exchanger is calculated, taking into account viscous dissipation along the plates and reverse heat conduction toward the stack. An optimal set of geometrical and relaxation parameters for the heat exchanger is obtained. Results are compared with experimental measurements of temperature fluctuations behind the stack plates, and with the results of numerical simulations from the literature. [This work is supported by ANR (project MicroThermoAc NT051$_4$2101).]

11:45

3aPAb7. Thermoacoustics and thermal dissociation of water. Gregory W. Swift and Drew A. Geller (M.S. K764, Los Alamos Natl. Lab., Los Alamos, NM 87545, swift@lanl.gov)

Near 2600 K, 10% of water molecules are thermally dissociated at atmospheric pressure, with a reaction time constant below 1 ms. Such temperatures can be reached with focused sunlight. To use this endothermic reaction for the production of hydrogen, the hydrogen must be separated from the oxygen at high temperature, because they would quickly recombine to form water again, if the unseparated mixture were simply returned to lower temperatures. We have considered thermoacoustic mixture separation for this purpose. Our calculations show that the thermal-diffusion ratios are high enough to yield steadily flowing streams of hydrogen-enriched steam and oxygen-enriched steam in a separation channel less than a wavelength long. However, the thermoacoustic power density in 1-bar steam is low, so the required apparatus would be large, needing alot of expensive and fragile high-temperature material, such as calcia-stabilized zirconia. Our estimates show that this approach to solar hydrogen production would be approximately 30 times more expensive than solar-Stirling electricity generation driving traditional water electrolysis. [Work supported by DOE Office of Science.]

Session 3aSCa

Speech Communication: Speech Intelligibility and the Vowel Space

Sarah Hargus Ferguson, Cochair
Univ. of Kansas, Dept. Speech Language Hearing Science and Disorders, Lawrence, KS 66045

Gary G. Weismer, Cochair
Univ. of Wisconsin, Waisman Ctr., Madison, WI 53705-2280

Chair's Introduction—8:00

Invited Papers

8:05

3aSCa1. Vowel space parameters. D. H. Whalen (Natl. Sci. Foundation, 4201 Wilson Blvd., Ste. 995, Arlington, VA 22230, and Haskins Labs.)

Vowels are universal in human languages and have long been categorized in feature systems. However, the feature systems are based on introspection and include a notion of ''height'' or ''closeness,'' that is not anatomically straightforward. From x-ray and other imaging data, the important aspects seem to be palatal, velar, and pharyngeal closures, with secondary effects of lip rounding. Describing vowels in terms of location of constriction makes some assimilation patterns more direct, as with palatal vowels (e.g., /i/) conditioning palatal fricatives. Vowels can also be described as intrinsically slower than consonants, which allows for a more straightforward description of their timing relationships, and a possible explanation for the crosslinguistic preference for CV syllables. The use of fricatives for syllabic nuclei is common in languages and derives easily from a narrowing of the constriction to the point at which frication is generated. More and less vowel-like patterns in consonants have also been noted, such as in coarticulation of laterals and rhotics. Gesture-based systems have difficulty describing some phonological patterns (e.g., vowel lowering). The status of schwa, which seems to lack a constriction, is unclear. With two competing systems having different advantages, work directly comparing the approaches is needed within speech science.

8:35

3aSCa2. Vowels and speech intelligibility. Diane Kewley-Port (Speech Psychophysics Lab., Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN 47405)

Recent research on the contribution of vowel versus consonant segments to sentence intelligibility has demonstrated a 2:1 advantage for vowels in a noise replacement paradigm [Kewley-Port et al., J. Acoust. Soc. Am. (in press)]. The vowel advantage is sustained when boundaries are slightly modified in sentences, but different results have been reported for isolated words. Our follow up research with typical elderly hearing-impaired listeners has demonstrated that the vowel segmental information is especially important in sentence intelligibility, even though it is commonly believed that consonants are more important than vowels for speech recognition. Additional research in our lab has examined the relation between listeners' ability to discriminate acoustic detail in vowel formants at a more peripheral level of the auditory system, and to identify vowel categories at more cognitive/linguistic levels. Three populations of listeners have participated in these experiments: Normal-hearing Americans, hearing-impaired Americans, and normal-hearing second language learners of English. Moderate correlations between formant discrimination and vowel categorization were observed for all three groups. Implications of our results will be discussed in terms of theories of speech perception and of second language speech learning. [Work supported by NIH.]

9:05

3aSCa3. The acoustic vowel space in speech disorders: Data and interpretation. Gary Weismer (Dept. of Communicative Disord. and Waisman Ctr., Univ. of Wisconsin-Madison, 1975 Willow Dr., Madison, WI 53706)

The acoustic vowel space, usually defined by the area enclosed by the corner vowels in an F1-F2 plot, may have some use as a general index of speech severity in various types of speech disorders. In this presentation, vowel space data will be reviewed for a number of different speech disorders, and the issue of the role of vowel space measurements in understanding both the underlying speech production deficit, as well as deficits in speech intelligibility, will be considered. Vowel space estimates, it will be argued, are in large part a gross estimate of speech motor control deficit. This is in contrast to a view of vowel space estimates as direct predictors of, say, specific aspects of segmental articulation difficulties or componential aspects of speech intelligibility deficits (for example, the extent to which vowel problems predict a speech intelligibility deficit). Some suggestions for future work, including extensions of the vowel space concept to include planar estimates of formant motion, will be presented. [Work supported by NIH.]

9:35

3aSCa4. Speech intelligibility and the vowel space: Normal speech. Amy T. Neel (Dept. of Speech and Hearing Sci., Univ. of New Mexico, MSC01 1195, Albuquerque, NM 87131-0001)

Vowel formant frequency values and related vowel space measures have been widely used in the study of speech intelligibility for normal and disordered talkers. The strength of vowel space measures in predicting speech intelligibility for normal talkers, however, is relatively low: [Bradlow et al. (1996)] found that vowel space dispersion predicted 19% of variance in sentence intelligibility, and [Hazan and Markham (2004)] found that the F2 difference between /i/ and /u/ predicted 16% of variance in word intelligibility scores. Using data from the 45 men and 48 women speakers in the Hillenbrand et al. (1995) database, acoustic characteristics of ten vowels were used to predict identification accuracy. Multiple regressions revealed that global measures (mean f0, F1 and F2, duration, and amount of formant movement) and fine-grained measures (vowel space area, mean distance among vowels, ranges for f0, F1, and F2, duration ratio between long and short vowels, and dynamic ratio between dynamic and static vowels) accounted for less than one-quarter of the variance in identification scores across talkers. Focusing on confusions among spectrally similar vowels may provide better information on intelligibility differences among normal talkers. Goodness ratings may provide a wider range of scores for statistical analysis than identification accuracy.

10:05–10:30 Break

10:30

3aSCa5. Clear speech effect on vowel production across languages. Rajka Smiljanic and Ann Bradlow (Linguist., Northwesten Univeristy, 2016 Sheridan Rd., Evanston, IL 60208 rajka@northwestern.edu)

In this talk, we discuss vowel production in clear speech, a distinct mode of speech production intended to enhance intelligibility, across languages with different phonological properties (Croatian and Spanish with small vowel inventories vs. English with a large vowel inventory). A comparison of Croatian and English revealed the equivalent clear speech vowel space expansion in both languages. In addition, listeners recognized words in noise more accurately in clear than in plain speech in their native language, establishing that the plain-to-clear speech articulatory modifications increased intelligibility. A comparison of Spanish and English also showed similar amounts of vowel space expansion as well as the maintenance of coarticulatory patterns in clear speech for both languages. The inventory-independent vowel space expansion in all three languages suggests that talkers hyperarticulate even when segments are unlikely to be perceptually confusable (few vowel categories that are fairly distinct). The maintenance of coarticulation further suggests that spreading of segment identity across neighboring sounds may be beneficial to the listener. In this talk, we also discuss various measures of vowel space expansion and different clear speech strategies across talkers. We conclude with some remarks about the link between the identified acoustic-phonetic features of clear speech and intelligibility.

11:00

3aSCa6. Listener influences on vowel intelligibility. Tessa Bent (Dept. of Psychol. and Brain Sci., Indiana Univ., 1101 E. 10th St., Bloomington, IN 47405, tbent@indiana.edu), Sarah Hargus Ferguson (Univ. of Kansas, Lawrence, KS 66045), and Diane Kewley-Port (Indiana Univ., Bloomington, IN 47405)

Do the acoustic-phonetic parameters that promote highly intelligible speech vary across different listener populations? The current study investigated whether inter-talker differences in American English (AE) vowel intelligibility were maintained across three listener groups: Normal hearing native AE-speaking adults (NH), normal hearing Korean-speaking adults learning AE as a second language (L2), and hearing impaired native AE-speaking adults (HI). These groups heard recordings of 10 AE vowels in /bVd/ context produced in conversational speech style by 12 talkers. The stimuli were mixed with noise and presented for identification in a 10-alternative forced choice task. Vowel intelligibility varied substantially among the talkers. There were significant correlations of talker scores for the NH and HI listeners, but not for NH and L2 listeners. An analysis of acoustic-phonetic vowel features indicated that vowel intelligibility was negatively correlated with vowel space expansion for the HI group, while for the L2 group, intelligibility was positively correlated with vowel duration. These data suggest that although both HI and L2 listeners have difficulty in noisy listening situations, their degraded performance has different underlying causes that lead to substantial differences in the perception of the intelligibility of different talkers. [Work supported by NIH-NIHDCD 02229 and T32-DC00012.]

Session 3aSCb

Speech Communication: Speech and Prosodic Issues (Poster Session)

Winifred Strange, Chair
CUNY Graduate School, Program in Speech and Hearing, 365 Fifth Ave., New York, NY 10016

Contributed Papers

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

3aSCb1. How did a stimulus order in AX discrimination training influence improvement in the ability to perceive an American English /l-r/ and /gl-gr/ among Japanese listeners? Teruaki Tsushima (Ryutsu-Kagaku Univ., 3-1, Gakuen-Nishi-Machi, Nishi-Ku, Kobe 651-2188, Japan, Teruaki_Tsushima@red.umds.ac.jp)

Previous research has shown that discriminability of /l-r/ in AX discrimination may be significantly lower when /l/ is presented as the first stimulus than otherwise, among Japanese listeners under certain experimental conditions. The present study examined whether the stimulus order in AX discrimination training had significant effects on improvements of the ability to discriminate and identify English /l-r/ and /gl-gr/. One group of Japanese listeners ($N=12$) received twelve sessions of AX training in which stimuli (i.e., lexical items) were presented in the order of /l/ to /r/ (e.g., glock-rockh), while the stimulus order was reversed (e.g., grock-lockh) in the other ($N=12$). A comparison of pre-test and post-test results found the former group showed a significant improvement of the ability to discriminate /l-r/ ($p=0.037$), to identify /l/ in /l-r/ ($p=0.023$), and a significantly better-defined category boundary of synthetic /la-ra/ stimuli than the other group in the post-test ($p=0.05$), while none of the improvements were significant in the other group. The results indicated, unlike what was predicted from the previous findings, the ability to perceive /l-r/ may be better improved when /l/ was presented first than otherwise during AX training. The results will be discussed in terms of models of L2 speech learning.

3aSCb2. Nonnative phonemes are open to native interpretation: A perceptual learning study. Matthias J. Sjerps and James M. McQueen (MPI for Psycholinguist., Postbus 310, 6500 AH, Nijmegen, The Netherlands, matthias.sjerps@mpi.nl)

Four experiments examined whether Dutch listeners can learn to interpret a nonnative phoneme (English [θ]) as an instance of a native category (Dutch [f] or [s]). During exposure in Experiment 1, two listener groups made lexical decisions to words and nonwords. Listeners heard [θ] replacing [f] in 20 [f]-final words (Group 1), or [s] in 20 [s]-final words (Group 2). At test, participants heard e.g., [doθ], based on the minimal pair doof/doos (deaf/box), and made visual lexical decisions to e.g., doof or doos. Group 1 were faster on doof decisions after [doθ] than after an unrelated prime; Group 2 were faster on doos decisions. The groups had thus learned that [θ] was, respectively, [f] or [s]. This learning was thorough: effects were just as large when the exposure sound was an ambiguous [fs]-mixture (Experiment 2) and when the test primes contained unambiguous [f] or [s] (Experiment 3). In Experiment 4, signal-correlated noise was used as the exposure sound. Listeners learned that the noise was an [f], irrespective of [f]- or [s]-biased exposure, showing that learning is determined by the new sounds spectral characteristics. Perceptual learning in a native language is thorough, and can override years of second-language phonetic learning.

3aSCb3. Effectiveness of a robust computer assisted pronunciation training tool. Kwansun Cho, John G. Harris (Dept. of Elec. and Comput. Eng., Univ. of Florida, Gainesville, FL 32611, kscho@cnel.ufl.edu), and Ratree Wayland (Univ. of Florida, Gainesville, FL 32611)

A reliable ASR-based pronunciation training tool named STAR (self-training accent reduction) is implemented for native speakers of Korean learning American English. STAR is designed to focus on the most frequent phonemic errors made by Korean adult learners and to provide instantaneous feedback. In order to investigate the effectiveness of STAR, ten Korean participants are recruited for this pilot study. The study consists of three phases: Pre-test, training, and post-test. During the pre-test, the participants read a pre-designed word list containing accent sensitive phonemes into a microphone and participate in one session of training using the STAR system. The participants return for two more sessions of training on the following day. The post-test is administered on the third day, after one additional training session. During the post-test, the participants read the same wordlist as the one administered during the pre-test, and an additional wordlist that they were not trained on. Two trained phoneticians listen and transcribe all recordings to examine whether the participants' productions are more accurate after the STAR training. The results indicate that most of the participants who practice pronunciation with STAR show improvement in their pronunciation.

3aSCb4. The adaptability of laboratory phonemic perception training protocols to common second language instruction situations. Thomas R. Sawallis and Michael W. Townley (English Dept., Univ. of Alabama, Box 870244, Tuscaloosa, AL 35487, tsawalli@bama.ua.edu)

For the past two decades, second language pedagogy has de-emphasized phoneme-level pronunciation training. During the same period, laboratory experiments have demonstrated a variety of benefits from training non-natives in the perception of difficult target language contrasts. Experiments have shown that learners' perceptual performance improves (Jamieson & Morosan, 1986; Flege, 1989 & 1995), that the improvements generalize to new talkers and words (Lively, Logan, & Pisoni, 1993), that perceptual training triggers production improvements (i.e., without specific pronunciation training, Bradlow et al., 1997), and that both perceptual improvements (Flege, 1995; Lively et al., 1994) and production improvements (Bradlow et al., 1999) continue over several months. Protocols for these training experiments are generally intensive, and may include sessions of 40 min, three sessions per day, or sessions for 15 consecutive business days. Such onerous protocols are acceptable for basic research with compensated subjects. They are unacceptable for normal L2 instruction, with paying learners and with limited time for pronunciation training. The present study analyzes the training protocols from selected research articles, and documents the types and amounts of training. It then suggests

acceptable alternative regimens for achieving comparable training in typical learning situations, including 16-week college semesters and shorter language institute terms.

3aSCb5. The categorical nature of tones and consonants: Evidence from second language perception and production. Yen-Chen Hao and Kenneth de Jong (Dept. of Linguist., Indiana Univ., 322 Memorial Hall, 1021 E. 3rd Str., Bloomington, IN 47405, yehao@indiana.edu)

Tones are commonly considered to be psychologically equivalent to phonetic segments like consonants and vowels [Ladd, Intonational Phonology (1996)]. However, comparing two current studies reveals intrinsic differences between tones and consonants. One study examines ten American learners acquiring Chinese tones. The other examines 40 Korean learners acquiring English obstruents. Both studies include three tasks: Identification—subjects identified the tones or consonants of L2 nonsense words. This task requires auditory perception and associating the sound with a linguistic category. Mimicry—subjects listened and repeated the stimuli. This task requires perception and production. Reading—subjects read a list of nonsense words. This task requires associating linguistic labels with speech production. These two studies yield different patterns. For tones, accuracy rates for Mimicry are the highest, suggesting learners have more problems with linguistic association, the component shared by Identification and Reading. For consonants, accuracy rates for Mimicry are the lowest, signifying difficulty with perception and production, but not with linguistic association. Taken together, these findings suggest a pervasive difference in categorical nature of tones and consonants. These results also highlight the necessity of multiple tasks in assessing how linguistic contrasts function in a phonetic system. [Work supported by the NSF.]

3aSCb6. Neutralization in the perception and production of English coda obstruents by Korean learners of English. Hanyong Park, Yen-chen Hao, and Kenneth J. de Jong (Dept. of Linguist., Indiana Univ., 332 Memorial Hall, Bloomington, IN 47405, hanypark@indiana.edu)

Phonological neutralization rules require the suspension of differences between segments in perception and production. This paper examines the role of neutralization in the production and perception of Korean learners of English. In Korean, laryngeal and manner contrasts found in initial obstruents are systematically neutralized in coda position into stops that sound like voiceless stops in English. The current paper pursues two questions: (1) Does neutralization have the same effect on perceptual and production abilities? (2) are neutralization effects found with all English segments, or are they restricted to stops, which are transferred from Korean? /p, b, t, d, f, v, θ, δ/ were placed after /a/. Two tasks are compared: (A) Identification. Forty Korean learners identified coda consonants produced by four native speakers of English, using English labels. (B) Reading. Ten native English listeners identified coda consonants produced by four Korean learners. Identification errors were largely unidirectional; for example, many segments were labeled /f/, while /f/ identification was highly accurate. Reading errors were largely bidirectional with a systematic tendency for more errors, resulting in voiceless consonants and stops, conforming to the Korean neutralization pattern. Such neutralization effects were as prevalent in fricatives as stops. [Work supported by NSF.]

3aSCb7. Perceptual learning about word boundaries with familiar and unfamiliar voices and accents. Rachel Smith (Dept. of English Lang., Univ. of Glasgow, 12 University Gardens, Glasgow G12 8QQ, U.K., R.Smith@englang.arts.gla.ac.uk)

An experiment investigated perception of speaker-specific variation in phonetic detail around word boundaries. Two male speakers of standard Southern British English (SSBE) read 24 phonemically-identical sentence pairs that differed in the location of a word boundary, e.g., *So he diced them* versus *So he'd iced them*. In pre- and post-tests, 80 SSBE subjects heard the sentences in cafeteria noise (SNR 2 dB) and typed the words they heard. Between these tests, subjects received 40 minutes' training: They heard different tokens of the same sentences without noise in a disambiguating context, and answered questions about their meaning. A counterbalanced between-groups design manipulated the voices heard in tests (speaker 1 or 2; always the same in pre- and post-test) and in training (Same as in tests, or Different). Training with the Same voice caused significantly more improvement in identification of words and syllable constituents at word boundaries than training with a Different voice. Preliminary data from 13 subjects with a different accent (Glaswegian Scottish) indicate, in contrast, more improvement after training with a Different voice. Apparently, when an accent is familiar, perceptual learning about word boundaries is voice-specific; when the accent is unfamiliar, learning generalizes to other speakers of the accent.

3aSCb8. Experimental paradigm influence subject's perception of attitudes. Caroline Menezes, Donna Erickson, Kikuo Maekawa, and Hideki Kawahara (NIJL, 3591-2 Midori-cho, Tachikawa-shi, Tokyo 190-8561, Japan; Showa Music Univ., Kawasaki City, Kanagawa ken, 215-8558, Japan; Wakayama Univ., Japan)

Japanese utterances spoken with different attitudes (admiration, suspicion, and disappointment) were independently manipulated for pitch-contour and voice quality using STRAIGHT {Kawahara}. Close copy stylization of the prototypical pitch contour for the three attitudes was imposed on the naturally spoken utterances, producing stimuli with all combinations of voice quality types and pitch-contour shapes. The utterances were submitted to two separate experiments, wherein subjects were asked to judge the attitude of the morphed utterances. The first, a forced choice experiment, asked subjects to choose if the utterances were admiration, suspicion, or disappointment, and the second, a free choice experiment, where subjects could freely choose the attitude they perceived. The results from the forced choice test indicated that subjects used pitch contour cues to choose speaker attitude. However, the results from the free choice paradigm indicated that subjects used both voice quality and intonation cues, and were able to perceive more affect types than the original three. Whereas the forced choice paradigm suggested that intonation was the primary cue for listeners to determine speaker attitudes, the free choice paradigm suggested an independence of voice quality and intonation in determining speaker attitudes, which resulted in a larger number of perceived attitudes by listeners.

3aSCb9. The role of prosodic boundaries in comprehension of Korean pseudo-cleft sentences. Jaehoon Jeong (Dept. of Linguist., Univ. of Hawaii, Honolulu, HI 96816, jhjeong@hawaii.edu)

A large prosodic boundary triggers, in general, a high attachment of an ambiguously attached phrase (e.g., Susie learned that Bill called // on Monday). The informative boundary hypothesis (IBH), however, claims that such an effect of a local prosodic boundary is neutralized by the presence of another boundary (e.g., Susie learned // that Bill called // on Monday). According to the IBH, it is not the absolute size of the later boundary (here, pre-PP boundary) but its size relative to any earlier boundary before a constituent containing only the lower attachment site. Since the IBH implies nothing about the interpretation of an ambiguous phrase whose attachment sites appear later, the current study, as an extension of the IBH, investigates whether the effectiveness of a prosodic boundary can be determined by its size relative to a relevant subsequent boundary. The results showed that the interpretation of the sentence-initial subject NP was affected by the absolute size of the post-NP boundary, but the later boundary type had nothing to do with the interpretation. This provides further evidence that prosodic information is processed in an incremental fashion (left-to-right) and perceivers tend to impose prosodic structures as fast as possible.

3aSCb10. On voicing activity under the control of emotion and loudness. Samuel Kim, Sungbok Lee, and Shrikanth Narayanan (Speech Anal. & Interpretation Lab. (SAIL), Univ. of Southern California, 3740 McClintock Ave., Los Angeles, CA 90089, kimsamue@usc.edu)

For a parametric study of interplay between loudness control and emotional modulation in voicing activity, electroglottography (EGG) data collected from two male and two female subjects are examined. The subjects read emotionally neutral sentences in a self-controlled manner with four different emotional states (neutral, angry, sad, happy) and three levels of loudness (soft, normal, loud). The analysis focused on the timing and shape-related EGG waveform parameters of the vowel /a/ in the data as a function of emotional state and loudness level. Specifically, open quotient (OQ), speed quotient (SQ), and noise-to-harmonic ratio (NHR) of EGG waveform are investigated. Pitch, root-mean-square (RMS) energy, and duration of speech waveform are also analyzed. Despite inter-subject differences, some general tendencies of the EGG parameters can be observed. It is found that angry emotion shows the lowest OQ in a given loudness level, and happy emotion exhibits significantly higher OQ than angry emotion. SQ mainly varies along the loudness dimension, smaller SQ in louder voice. NHR is the highest in soft voice and shows no clear emotion-dependent pattern. The results suggest that OQ is the main EGG parameter that is controlled by speakers for emotion expression, in addition to its role in loudness control.

3aSCb11. Intraglottal pressures in a static physical model of the uniform glottis: Entrance loss coefficients and viscous effects. Lewis Fulcher and Ronald Scherer (Phys. and Astron. and Commun. Disord., Bowling Green State Univ., Bowling Green, OH 43403, fulcher@bgsu.edu)

Pressure distributions for the uniform glottis were taken with a static physical model (M5) for the diameters $d=0.005$, 0.0075, 0.01, 0.02, 0.04, 0.08, 0.16, and 0.32 cm for a number of transglottal pressures of interest for phonation. At each pressure and diameter, entrance loss and exit coefficients are calculated. The pressure dependence and the diameter dependence of these coefficients are catalogued and compared with some standard values from the earlier literature. The accuracy with which tabulations of these coefficients reproduce the M5 pressure distributions is examined. To an excellent approximation, the intraglottal pressures at smaller diameters decrease linearly with the axial distance, and remnants of this behavior are seen at $d=0.08$ cm and 0.16 cm. It is shown that the intraglottal pressure gradients are linear functions of the glottal flow rates. Thus, dividing the pressure gradients by the flow rates allows one to isolate the geometric dependence of viscous effects. It is shown that an inverse five-halves power law for the diameter dependence of the viscous glottal resistance is a better approximation than the inverse cube law proposed by van den Berg, Zantema, and Doornenbal. [Work supported by NIH R01DC03577.]

3aSCb12. The effects of linguistic experience on the perception of pathologically-disordered phonation. Christina Esposito (Macalester College, 1600 Grand Ave., St Paul, MN 55105)

Ladefoged [Vocal Fold Physiology: Contemporary Research and Clinical Issues, College Hill Press, San Diego, 1983] wrote, what is a pathological voice quality in one language may be phonologically contrastive in another If one persons pathology is another persons phonemic phonation, then speakers of languages with phonation contrasts should treat pathologically-disordered and linguistically-relevant phonations in the same way. To test this, Gujarati (breathy versus modal contrast), Spanish (no breathiness) and English (allophonic breathiness) listeners participated in similarity-rating tasks. One used Mazatec (breathy, modal, and creaky phonation) stimuli, the other pathologically-disordered stimuli. Gujaratis treated pathologically-disordered stimuli and Mazatec stimuli in the same way. With both types of stimuli, Gujaratis based their judgments on H1$-$H2 (amplitude of the first harmonic minus the second) and mapped the stimuli into clusters with similar H1$-$H2 values across the two experiments. In both experiments, English listeners relied on cepstral peak prominence and H1$-$H2, and Spanish listeners relied on H1$-$H2 and H1$-$A1 (amplitude of first formant peak). English and Spanish listeners grouped both types of stimuli into clusters with similar values for the acoustic measures. However, English and Spanish listeners differed in how they weighed the importance of the different dimensions across the experiments. To English and Spanish listeners, pathologically-disordered phonations are subtly different from linguistically-relevant ones.

3aSCb13. Analysis of voice perturbations using an asymmetric model of the vocal folds. Marco Nardone, Lewis Fulcher, and Ronald Scherer (Phys. and Astron. and Commun. Disord., Bowling Green State Univ., Bowling Green, OH 43403, mnardone1@verizon.net)

A mathematical model was developed to investigate possible causes of jitter and shimmer. The model builds on the classic, lumped element model of Ishizaka and Flanagan and allows for asymmetric motions of the vocal folds and aerodynamic imbalances. The intraglottal pressures were derived from empirical pressure data obtained from a static physical model of the larynx (M5). The mathematical model is based on ten, second-order, nonlinear, coupled, ordinary differential equations that were solved simultaneously using the software Mathematica. The solutions were analyzed graphically and numerically to identify perturbations in the fundamental frequency and amplitude of the glottal airflow. Jitter and shimmer were quantified using the jitter factor and the amplitude variability index. The results indicate that only time-dependent variations in biomechanical and aerodynamic parameters result in jitter and shimmer. The magnitudes of jitter and shimmer tend to be less than those observed in the natural sounding voice, even when the asymmetries are large. Although time-independent asymmetries may cause the vocal folds to oscillate out of phase or with different amplitudes, they tend to entrain and vibrate at a common frequency. [Work supported by NIH R01DC03577.]

3aSCb14. Phonetic correlates of phonological register in Takhian Thong Chong. Christian DiCanio (Dept. of Linguist., Univ. of California, Berkeley, 1202 Dwinelle Hall, Berkeley, CA 94720-2650)

The author's phonetic fieldwork from the Takhian Thong Chong language demonstrates that the four phonological registers are distinguished phonetically with distinct voice quality (phonation) contours, pitch contours, and durational properties. Both EGG and acoustic data were gathered from seven native speakers. A dynamic open quotient contour extracted from electroglottographic recordings demonstrates register-specific phonation type correlates. The four distinct registers are characterized by: Modal phonation throughout with a mid-level pitch, a contour from modal to tense phonation with a high rising and falling pitch contour, a contour from breathy to modal phonation with a mid-falling pitch contour, and a contour from breathy to tense phonation with a midrising and falling pitch contour. Investigations into the correlation between OQ (open quotient) values and pitch indicate a positive global correlation between the relative length of the glottal cycle and the open period, but a lack of correlation between dynamic OQ and pitch curves. Aside from providing a phonetic description of a typologically rare phonological contrast, these findings suggest that there is not a strong relationship between absolute changes in degree of glottal abduction and changes in pitch, contrary to previous claims (Silverman 1997). [Work supported by UC Berkeley Graduate Division.]

3aSCb15. Response bias, type and token frequency, and prosodic context in segment identification. Noah Silbert and Kenneth de Jong (Dept. of Linguist., Indiana Univ., Memorial Hall, Rm. 322, 1021 E. Third St., Bloomington, IN 47401)

Frequency effects are well documented in a variety of linguistic domains. However, the relationship between response bias and frequency in segment identification has received little attention. We hypothesize that

listeners are biased toward higher frequency segments. In order to evaluate this hypothesis, maximum likelihood response bias parameters from the similarity choice model (Shepard, 1957; Luce, 1963) were obtained for a number of previously published segment identification data sets and rank-order correlated with a number of type and token frequency measures calculated from the Hoosier Mental Lexicon. Results indicate that bias tends to correlate more highly with type than with token frequency. Furthermore, in native and non-native individual listener data [Cutler et al. (2004)], correlations are slightly higher between bias and prosodically conditioned frequency than between bias and position-independent frequency, and correlations between bias and coda frequency are large and positive, whereas bias is not correlated with onset frequency. In addition, coda and position-independent measures of (type) frequency correlate more highly with one another than either do with onset measures. Accuracy variation and range of frequency distributions do not account for these effects. No other explanations for the observed prosodic differences are apparent. [Work supported by the NSF.]

3aSCb16. Model-based quantification of pathological voice production. Raphael Schwarz, Dimitar Deliyski (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, 1621 Greene St., Columbia, SC 29208), Joerg Lohscheller, and Michael Doellinger (Univ. Hospital Erlangen, 91054 Erlangen, Germany)

Hoarseness, the primary symptom of voice disorders, results from irregular vocal fold vibrations. The oncological therapy of laryngeal cancer may even result in a total loss of voice if an excision of the larynx, and thus, the vocal folds, is necessary. State-of-the-art voice rehabilitation technique in this case is the utilization of scarred tissue in the upper part of the esophagus for substitute voice production. The quality of laryngeal voice, as well as the substitute voice, primarily depends on the anatomy and the vibration patterns of the voice-producing element. Using endoscopic high-speed recordings, the voice generators are observed during voice production. In this work, a model-based approach feasible for the analysis and objective quantification of vocal fold vibrations, as well as the PE dynamics, is presented. By means of an automatic parameter optimization, the dynamic of a biomechanical model of the considered voice-producing element is fitted to the recorded vibration patterns. Thereby spatial and temporal properties of the vibrations are incorporated. The resulting values of the optimization parameters represent an objective quantification of the vibration patterns. In addition, the model parameters enable an approximation of physiological tissue parameters as stiffness and mass distribution.

3aSCb17. Neural representation of pitch-shifted voice feedback at N100 and P200 components: An event-related potentials (ERP) study. Roozbeh Behroozmand, Hanjun Liu, and Charles R. Larson (Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60208)

Numerous studies have shown that perturbations in voice pitch feedback lead to compensatory responses in voice fundamental frequency (F0), but few have reported on neural responses to the acoustical stimulation during the pitch-shifted voice feedback. In this study, randomized-onset, upward pitch-shift stimuli (60 ms duration) were presented to ten subjects during sustained vowel phonation while the event-related potentials (ERPs) were recorded through Cz-linked earlobe surface electrodes. ERPs were also recorded as subjects passively listened to the same feedback signal that was recorded during vocalization. The experimental task was repeated for 100, 200, and 500 cents conditions, respectively, and the results were compared in vocalization and listening conditions. Across three stimulus magnitudes, the N100 responses during vocalization were suppressed in comparison with those recorded in the passive-listening condition. On the other hand, the P200 responses in the passive-listening task were graded in amplitude according to the magnitude of the stimulus, with the smallest responses for 100 cents and largest for 500 cents stimuli. However, during vocalization, the P200 responses were found to be graded in the opposite way, where the largest responses were associated with 100 cents and the smallest with 500 cents stimuli.

3aSCb18. Detecting rhythmical prominence in speech by an optimized convolution kernel. Christina Orphanidou and Greg Kochanski (Phonet. Lab., Univ. of Oxford, 41 Wellington Sq., Oxford OX1 2JF, UK)

We present an approach for detecting rhythmical prominence in read speech. A production experiment was conducted during which subjects repetitively read out speech to a metronome, trying to match stressed syllables to its beat. In the analysis, we compute a function from the speech waveform, related to acoustic properties of speech such as specific loudness, pitch, voicing, and spectral slope. The function is then convolved with a Mexican Hat convolution kernel. Taking large maxima in the function to be predictions of the metronome ticks, we adjust the parameters of the signal to maximize the accuracy of the predictions. The parameters are adjusted by minimizing the phase variation between metronome ticks and ticks predicted from the audio, over a specified time interval. We confirm the results by Bootstrap resampling. We find that the most important factor is the contrast in specific loudness between a syllable and its neighbors. The prominence can be deduced from the specific loudness in an (approximately) 360 ms window centered on the syllable in question relative to an (approximately) 800 ms-wide symmetric window.

3aSCb19. The influence of position-in-utterance on the scale of articulatory gestures. Christina Kuo and Gary Weismer (Dept. of Communicative Disord. and Waisman Ctr., Univ. of Wisconsin-Madison, 1975 Willow Dr., Madison, WI 53706)

The influence of prosodic structures of utterances on articulation has been of interest in speech production. Data on fundamental frequency, peak oral airflow, movement, and speech timing has been reported. Nonetheless, little is available in the literature regarding the magnitude of articulatory behavior as a prosodic phenomenon. This study is motivated by the phenomenon of articulatory declination, which is a hypothesized gestural weakening over an utterance, analogous to the better known F0 declination. An attempt is made to describe and examine articulation at the start and the end of an utterance under varied stress during sentence reading of normal speakers aged 20 to 30. Target words were ''row,'' ''sew,'' and ''sigh.'' A control word ''bee'' was included given the diphthongal nature of the three targets. Each target word was surrounded by the neutral vowel ə and embedded in a grammatical sentence under five position-in-utterance conditions: initial, initial-stressed, final, final-stressed, and final without a following ə. Formant transitions were measured from the first glottal pulse after the consonant through the following ə. Transition extent and duration were derived for analysis. Pilot data with two participants (one male, one female) and work in progress are discussed within the framework of articulatory declination.

3aSCb20. Gradiency and categoricity in prosodic boundary production and perception. Jelena Krivokapic (Dept. of Linguist., Univ. of Southern California, 3601 Watt Way, GFS 301, Los Angeles, CA 90089-1693) and Sankaranarayanan Ananthakrishnan (Univ. of Southern California, Los Angeles, CA 90089)

A study investigating whether the production and perception of prosodic boundaries is categorical or gradient is presented. Most theories assume a small set of prosodic categories which are marked by categorically different prosodic boundaries (e.g., Beckman & Pierrehumbert 1986 Phonology Yearbook, 3, 255–309, Nespor & Vogel 1986 Prosodic Phonology), while an alternative view suggests the possibility of gradiently varying prosodic boundaries (e.g., Byrd & Saltzman, 2003 J. of Phonetics, 31, 149–180). The first part of this study is an articulatory study investigating the production of twenty-four prosodic junctures ranging from no

boundary to very strong boundary. In the second part of the study listeners evaluate the perception of these same boundaries. Two evaluations (histograms and fitting mixture distributions) were conducted on both the production and on the perception data. The production of prosodic boundaries was found to be categorical, showing a large (IP) and a predominantly small prosodic boundary. The perception of prosodic boundaries showed that listeners perceive two distinct categories, but also that the data is better explained if more prosodic categories, up to 8, are assumed. [Supported by NIH.]

3aSCb21. Discrimination of lexical tone contrasts in disyllabic nonsense pairs by adult speakers of Mandarin and of English. Shari Berkowitz and Winifred Strange (CUNY Grad. Ctr., Speech Acoust. and Percept. Lab, 365 Fifth Ave., New York, NY 10016, sberkowitz1@gc.cuny.edu)

Most previous research on cross-language perception of lexical tone has used monosyllabic stimuli. In running speech, coarticulation affects the surface forms of the tones, creating contours that may be much more difficult to discriminate. Two tokens per type of disyllabic Mandarin nonsense words served as stimuli in a categorial same/different task. Pairings were chosen to be challenging, with some easy control pairs. Native Mandarin listeners ($N=5$) and American English listeners ($N=15$) with no background in tone languages completed 224 trials with feedback on 14 contrasts, then completed two additional blocks of 224 trials without feedback. Mandarin speakers were at ceiling on all contrasts; for Americans, accuracy did not improve across training and test blocks. Overall accuracy for different pairs averaged 78% (range 54–99%), and overall accuracy for (categorial) same pairs averaged 84% (range 75–97%). The mean A' score for Americans was 0.88 (range 0.75-0.96). Performance across the 14 contrasts varied from $A'=0.74$ to 0.96, and showed significant context effects. Error analysis showed a pattern of errors across tone pairs that supported the hypothesis that American speakers relied on the overall pitch contour across the two syllables, rather than a syllable-by-syllable analysis.

3aSCb22. An exemplar-based approach to automatically detect burst in word-initial voiceless stops in spontaneous speech. Yao Yao (Univ. of California, Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, yaoyao@berkeley.edu)

This paper discusses a methodological question: How to automatically extract phonetically important but untranscribed information (such as VOT and closure duration) from a large spontaneous speech corpus? In this study, we made use of the exemplar-based similarity scores, first developed in Johnson 2006, to find the point of burst in voiceless stops in the Buckeye speech corpus. A similarity score is in essence a measure of how similar one piece of acoustic information is to another on the spectrogram. When used together with phonal spectral templates of the speaker, it can be used to measure how similar a piece of acoustic data is to, say, the typical [aa], or the typical [f] of the same speaker. We find that a two-dimensional score vector (<silence> score plus <sh> score) is already adequate to recognize burst in stops and the pattern is robust enough across a wide range of context in uncontrolled spontaneous speech. It works confidently on average around 90% of the time; the mean error is estimated to be around 3–5 ms (the optimal error in theory is 2.5 ms due to the 5 ms step size in similarity scores). Its robustness, accuracy, and efficiency is examined in detail in the paper.

3aSCb23. Listener sensitivity to spectral slope attributes. Jody Kreiman and Bruce R. Gerratt (Head/Neck Surgery, UCLA School of Medicine, 31-24 Rehab Ctr., Los Angeles, CA 90095-1794)

Researchers have long known that the shape of the vocal source spectrum is an important determinant of vocal quality, but details regarding the perceptual importance of individual spectral features remain unclear. This study provides preliminary evidence about the perceptual importance of four acoustic source features: H1-H2 and the spectral slopes from 1.52 kHz, from 24 kHz, and above 4 kHz. Vowel stimuli were synthesized by varying each spectral parameter in steps. Because the perceptual salience of source parameters depends on F0 and on the inharmonic source spectrum, different series were created for a male and a female speaker, and for a source with little versus a large amount of inharmonic energy. Listeners heard all possible pairs of voices within each series and were asked whether stimuli were the same or different. We hypothesize 1) that listeners sensitivity to H1-H2 will greatly exceed their sensitivity to other parameters; and 2) that listeners sensitivity to H1-H2 and the slope of the spectrum from 1.52 kHz will be independent of noise, but that sensitivity to changes in the spectral shape above 2 kHz will depend on the presence of noise excitation in the voice.

3aSCb24. Robust and accurate F0 estimation for reverberant speech by utilizing complex cepstrum analysis. Masashi Unoki and Toshihiro Hosorogiya (School of Information Sci., JAIST, 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan)

This paper presents comparative evaluations of 12 typical methods of estimating the fundamental frequency (F0) over huge speech-sound datasets in reverberant environments. They involve several classical algorithms such as cepstrum, AMDF, LPC, and autocorrelation methods. Other methods involve a few modern algorithms, i.e., instantaneous amplitude and/or frequency-based algorithms, such as TEMPO, IFHC, and PHIA. The comparative results revealed that the percentage of correct rates and SNRs of the estimated F0s were reduced drastically as reverberation time increased. This paper, thus, proposes a method of robustly and accurately estimating F0 in reverberant environments by utilizing the MTF concept and the source-filter model in complex cepstrum analysis. The MTF concept is used in this method to eliminate dominant reverberant characteristics from observed reverberant speech. The source-filter model is used to extract source information from the processed cepstrum. Finally, F0s are estimated from them by using the comb-filtering method. Additive-comparative evaluation was carried out on the proposed method and other typical methods. The results demonstrated that it was better than the previously reported methods in terms of robustness and in providing accurate F0 estimates in reverberant environments. [Work supported by a Grant-in-Aid for Science Research from the Japanese Ministry of Education No. 18680017.]

Session 3aUW

Underwater Acoustics: Time Reversal and Matched Field Processing

Claire Debever, Chair
Scripps Inst. of Oceanography, Marine Physics Lab., La Jolla, CA 92093-0238

Chair's Introduction—8:00

Contributed Papers

8:05

3aUW1. Broadband, coherent inter-array processing of horizontal arrays. Claire Debever and W. A. Kuperman (Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., Mail Code 0238, La Jolla, CA 92093-0238, cdebever@ucsd.edu)

We study source localization in a shallow water environment using broadband data recorded on two horizontal arrays lying on the sea floor, each of length 250 m. They each had about 25 elements and were approximately 3.5 km apart [N. O. Booth et al., IEEE, JOE, **25**(3), (2000)]. The data on the arrays were produced by a source tow covering ranges of about 1 km to 10 km emitting tones from 50 Hz to 400 Hz. The data were processed using various conventional and adaptive algorithms for all the cases covering incoherent frequency and incoherent inter-array processing to coherent frequency and coherent inter-array processing, as a function of source position relative to the arrays. The results are compared to simulations. [Work supported by ONR.]

8:20

3aUW2. Bistatic target detection using a time-reversal operator. Geoffrey F. Edelmann, David M. Fromm, and Charles F. Gaumond (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave., Code 7145, Washington, DC 20375)

Previous work has shown that time-reversal operator (TRO) methods can detect ocean targets using a time-reversal mirror [C. Gaumond, J. Acoust. Soc. Am **119**, 976–990 (2006)]. To increase the signal-to-noise ratio, the TRO was measured by transmitting a set of orthogonal beams. The method is extended to a bistatic case where the receiver and transmitter are no longer collocated. Simulated deep ocean convergence zone and up-slope detection at 200 Hz will be discussed. The consequences of array geometry (horizontal and vertical) to beam orthogonality will be addressed. Specifically, under what conditions will a horizontal-receive array have enough aperture to measure a TRO of sufficient rank? It will be shown that the system must be able to both transmit and receive beams that are orthogonal with respect to the waveguide (and not simple free space arguments). [Work supported by the Office of Naval Research.]

8:35

3aUW3. Extraction of a backscattered target signature in a shallow water waveguide with decomposition of the time reversal operator method. Franck Philippe, Claire Prada, Julien de Rosny, Dominique Clorennec, and Mathias Fink (LOA, ESPCI, 10 rue Vauquelin, 75005 Paris, France)

In a shallow water waveguide, detection and characterization of a target is an active field of research. Retrieving the target signature is extremely difficult due to the complex mode coupling between the incident wave and the backscattered wave. One solution consists of deconvolving the target response from the waveguide impulse response. However this method needs an extensive a priori knowledge of the waveguide geometry [Mignerey *et al.*, JASA 1992 and Yang *et al.*, JASA 1994]. To overcome this difficulty, we propose to apply the DORT method (French acronym for decomposition of the time reversal operator). This method is based on time-reversal invariance. Time reversal is commonly known to be an effective means to focus on a target without any a priori knowledge of the propagation medium. In the same way, here we show that the eigenvalues of the time reversal operator are directly proportional to the backscattered target signatures, and the waveguide geometry only acts on the proportionality factor. A theory based on modal waveguide analysis is presented. Ultrasonic scale experiments prove the effectiveness of this method.

8:50

3aUW4. Effect on surface ship interference on adaptive beamforming for the shallow water array performance (SWAP) project. Richard L. Campbell, Jr. and Lisa M. Zurk (NEAR-LAB, ECE Dept., Portland State Univ., PO Box 751, Portland, OR 97207-0751, rlcamp@pdx.edu)

The shallow water array performance (SWAP) project is designed to explore the limits of large-aperture passive sonar array processing capability in a shallow-water environment with moving surface ship interference. The array, off the eastern coast of Florida near Ft. Lauderdale, has 500 elements with a total length of approximately 900 m. One challenge for the processing is the length of observation time needed by traditional adaptive beamforming formulations—with an array of this size and element quantity, the resolution of range and bearing cells is such that a ship may move across many cells during the snapshot time, spreading the resulting eigenvector structure and decreasing effective signal gain. A central question is the trade-off between array gain and this eigenvector spreading loss. To explore this question, tracks from actual ships in the vicinity of the array site, combined with sound speed and bathymetry data from the site, are used in an adiabatic normal mode simulation to predict the acoustic response across the array. The resulting simulated snapshots are used in adaptive and non-adaptive formulations to predict target detection performance as a function of the interference environment and processing parameters, for both full and sub-aperture processing schemes.

9:05

3aUW5. Source localization and interference suppression using mode space estimation. Kyungseop Kim, Woojae Seong (Dept. of Ocean Eng., Seoul Natl. Univ., Seoul 151-742, Korea), and Seongil Kim (Agency for Defense Development, Jinhae, Korea)

Weak target detection and localization in the presence of loud surface ship noise is a critical problem for matched field processing in shallow water. For stationary sources, each signal component of a received signal can be separated, and interference can be suppressed using eigen-space analysis schemes. However, source motion, in realistic cases, causes spreading of signal energies in their subspace. In this case, eigenvalues of target and interference signal components are mixed and hard to be separated with usual phone-space eigenvector decomposition (EVD) approaches. Our technique is based on mode space and utilizes the difference

in their physical characteristics of surface and submerged sources. Performing EVD for modal cross-spectral density matrix, interference components in the mode amplitude subspace can be classified and eliminated. This technique is demonstrated with data obtained from an L-shaped (vertical and horizontal) line array during the MAPLE IV experiment conducted in the East Sea, Korea, and results will be discussed.

9:20

3aUW6. Depth dependence of planewave beamformed data with a horizontal array in a waveguide. Kevin Cockrell and Henrik Schmidt (Dept. of Mech. Eng., Massachusetts Inst. of Tech., Rm. 5-204, 77 Massachusetts Ave., Cambridge, MA 02139, cockrell@mit.edu)

Despite extensive effort by the ocean acoustics community to design signal processing techniques that take advantage of the waveguide structure of ocean environments (e.g., matched field processing), planewave beamforming remains the only technique robust enough to be used with autonomous vehicles processing data in real-time. However, the waveguide affects the beamformed data in ways that can degrade the performance of bearing estimation. Precisely how the waveguide affects the beamformed data depends on several parameters: The acoustic source's range, the acoustic source's depth, the environmental parameters, and the horizontal array's depth. All of these parameters are "out of our control" except the depth of the towed array. Thus, numerical simulation is used to determine if there is an optimal depth to tow the horizontal array, which will minimize degradation to bearing estimation. [Work sponsored by ONR.]

9:35

3aUW7. Target scattering waves emitted from a time reversal array in shallow water. Yoshiaki Tsurugaya (NEC Corp., 1-10, Nissin-cho, Fuchu, Tokyo 183-8501, Japan, y-tsurugaya@bp.jp.nec.com), Toshiaki Kikuchi (NDA, Yokosuka, Kanagawa 238-0024, Japan), and Koich Mizutani (Univ. of Tsukuba, Tsukuba, Ibaraki 305-8571, Japan)

When a target exists between a sound source and a TRA in shallow water, the TRA receives the sound waves from the sound source and the waves scattering by the target. If the time reversal processing to them is carried out and they are re-emitted from the TRA, it will be thought that they are converged on the position of the sound source and the target. However, since the waves converging on the sound source have a high level, the waves converging on the target position are usually masked by the high level sounds. We cannot observe the waves converged on the target. We then eliminate only the high level sounds from the sound fields. In each array element, the signals in the case of nontarget are subtracted from the signals including the target. As a result of subtraction, the components of the scattering wave by the target are left on the array elements. The time reversal fields of the scattering wave are constructed by radiating the components of the scattered wave from each element again.

9:50

3aUW8. Identification of a resonant target buried in sediment using iterative, single channel, time reversal. Zachary J. Waters, R. Glynn Holt, Ronald A. Roy (Dept. of Aerosp. and Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, ronroy@bu.edu), and Benjamin R. Dzikowicz (NSWC-Panama City, Panama City, FL 32407)

The presence of noise and clutter makes identification of targets buried beneath the seafloor a challenging problem. Iterative time reversal using a single-channel transducer is shown to enhance the signal-to-noise ratio of backscattered echo returns from a buried resonant target. Each iteration consists of: (1) Insonifying the target with a broadband pulse, (2) windowing a portion of the backscattered echo return, (3) reversing it in time, and (4) using this waveform as the source signal for the subsequent interrogation. Scaled laboratory experiments are performed with a broadband ($rmQ\sim2$) transducer, operating between 500 kHz and 2 MHz, and a 6.35 mm diam hollow aluminum spherical shell target buried beneath a layer of simulated sediment. When the target is located within the time reversal window, there is a rapid convergence to a narrowband signal characteristic of a dominant mode of the target's scattering response. Images are generated by scanning the transducer laterally in two dimensions above the buried target. The images reveal enhancement of different resonant modes of the target, depending on the transducer's position. Results will also be reported from larger-scale experiments performed in a test pond at lower frequencies. [Work supported by Office of Naval Research Award No. N000140610044.]

10:05

3aUW9. Pressure sensitivity kernels applied to time-reversal acoustics. Kaustubha Raghukumar, Bruce D. Cornuelle, William S. Hodgkiss, and William A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238)

Time-reversal has been found to be more robust to sound speed perturbations than a one-way transmission. Various explanations have been advanced to account for the lower sensitivity of time-reversal. In this contribution, the robustness of time-reversal is quantitatively examined using sensitivity kernels. A first-order Born approximation is used to obtain the pressure sensitivity of the received signal to small changes in medium sound speed. The pressure perturbation to the received signal caused by medium sound speed changes is expressed as a linear combination of single-frequency sensitivity kernels weighted by the transmit signal in the frequency domain. This formulation can be used to predict the response of a source transmission to sound speed perturbations. The stability of time-reversal is studied and compared to that of a one-way transmission using sensitivity kernels. In the absence of multipath, a reduction in pressure sensitivity using time-reversal is only obtained with multiple sources. This can be attributed both to the presence of independent paths and to cancellations that occur due to the overlap of sensitivity kernels for different source-receiver paths. [Work supported by ONR.]

Session 3pAA

Architectural Acoustics: Acoustics of Rehearsal Facilities

Damian J. Doria, Chair
Artec Consultants, 114 W. 26th St., 12th Floor, New York, NY 10001

Invited Papers

1:00

3pAA1. Acoustics of university rehearsal spaces: Not too live, not too dead. Russell Cooper (Jaffe Holden Acoust., 114A Washington St., Norwalk, CT 06854, rcooper@jaffeholden.com)

The design of music rehearsal spaces for university schools of music must balance many factors. Most important of these is loudness and decay time. Rooms must have sufficient floor area and volume so that the rooms are not overly loud for the teachers and students. The size and volume of the room depend on the type of ensemble rehearsing in the room (marching band, concert band, wind ensemble, choral, orchestra, percussion, jazz/rock, etc.). Decay times are often desired to match that of the performance stage. We have found this not to be ideal, that in fact rooms should be less reverberant than the stage platform, to provide students and teachers the ability to hear their instrument's articulation and syncopation. A certain amount of adjustable acoustics are necessary to account for multi-use of the room for different ensembles, and finishes should be selected for the proper diffusion of sound and to achieve the desired decay time. This paper will discuss many of our firm's designs and some of the successful common design themes we use.

1:25

3pAA2. Straying outside the rehearsal room box. David Conant and William Chu (McKay Conant Brook Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362)

Acoustical considerations are seldom the singular driving factor in rehearsal room design. The desire in academia, to maximize use of precious construction footprint by accommodating an increasing variety of programs, while simultaneously adjusting to ever-decreasing budgets, has resulted in the emergence of hybrid rehearsal spaces that defy conventional classification. The described spaces by MCB serve musical groups with divergent acoustical demands and are variously, yet successfully, pressed into service for lectures, recitals, recording studios, and even experimental theater.

1:50

3pAA3. School music rehearsal spaces—How dry is dry enough? Bennett M. Brooks (Brooks Acoust. Corp., 27 Hartford Turnpike, Vernon, CT 06066)

Handbook values for optimum reverberation time (RT60) in a music studio, such as a school band or chorus rehearsal space, fall into the 0.8 to 1.0 second range, depending on volume. One music room furniture manufacturer specifies RT60 values as high as 1.91 seconds. We submit that these RT60 values are excessive. The Classroom Acoustical Standard (ANSI S12.60-2002) specifies a maximum allowable RT60 of 0.7 seconds to achieve good speech intelligibility. Obviously, good speech communication is necessary for effective education of any kind. In special function rooms designed for music instruction, experience has shown that the design target RT60 should be as low as 0.2 seconds, for frequencies of 500 Hz and above. The requirement for such low reverberation is dictated by the need to reduce the extremely high sound levels that can be generated by a large group; and to give the instructor a dry environment that provides the ability to isolate, identify, and evaluate the musical performance of an individual student while rehearsing in a larger group. This allows the instructor greater opportunity to reinforce good student musicianship and to correct individual student mistakes, ultimately resulting in a higher level of concert performance. Sonic examples are given.

Contributed Papers

2:15

3pAA4. Remedial acoustical design of a college band instrument rehearsal room modified because of asbestos treatment. Steven D. Pettyjohn (The Acoust. & Vib. Group, 5700 Broadway, Sacramento, CA 95820-1852, spettyjohn@acousticsandvibration.com)

Removal of asbestos material from an instrument rehearsal room at CSU, Sacramento, including acoustical treatment led to a new design. The fix dramatically altered the characteristics of the room and resulted in unacceptable conditions. Temporary installation of heavy velour curtains made the room tolerable, but a permanent fix was sought. The room has nonparallel surfaces, large and long, barrel-shaped diffuser panels, hard walls and floor, and a mostly hard ceiling. Field measurements were made under three conditions to quantify the room's acoustical conditions, including reverberation time and background sound levels. One-third octave band measurements were made from 12.5 to 10,000 Hz to cover all instruments and background sources. The 1-inch acoustical treatment first added to the room was not adequate to attenuate sound below 500 Hz, and insufficient to handle the high acoustical energy produced by the band. The echo between the floor and ceiling was very distinct, as neither surface had any treatment. Calculations using two different methods showed that substantial low-frequency sound absorption was required to reduce the sound at all frequencies to the design goal. The room is currently undergoing the modification, and results may be available at the time of the presentation.

2:30

3pAA5. Improvement of reverberation and sound isolation in a high school music rehearsal facility. Clemeth L. Abercrombie (Daly-Standlee and Assoc., 4900 SW Griffith Dr., Ste. 216, Beaverton, OR 97005, cabercrombie@acoustechgroup.com)

Music rehearsal facilities are used by some of the most acoustically demanding individuals; extremely discerning even at an early age in their musical path. This paper examines acoustical issues in a high school marching and concert band rehearsal suite that is home to approximately 70 students. Even though the two-year-old building met acoustical design goals provided by an acoustical consultant, the music program director using the facility was concerned enough with acoustical performance that he successfully lobbied the school district for a renovation. Outstanding issues included clarity and low-frequency response in the main rehearsal room, sound isolation and loudness in individual practice rooms directly adjacent to the main room, and noise experienced in the nearby choir rehearsal suite. This presentation will review the importance of acoustical parameters specific to rehearsal, as well as the implementation of custom designed low-frequency absorption and cost-effective sound-isolation improvements.

2:45

3pAA6. Concert hall canopy height comparison and computer model resolution efficacy. Michael Ermann and Indhava Kunjara Na Ayudhya (Virginia Tech. School of Architecture + Design, 201 Cowgill Hall (0205), Blacksburg, VA 24061, mermann@vt.edu)

This line of inquiry is twofold: (1) It uses computer modeling to compare the height of an adjustable concert hall ceiling canopy with established room acoustics metrics, and (2) It does so simultaneously with two computer models, one at a rough resolution, and one at a resolved resolution. The height comparison looks to establish best practices in operation of such a canopy in a hall; how high should it be set? The model resolution comparison looks to establish best practices in room acoustics computer modeling; how detailed must the model be?

THURSDAY AFTERNOON, 29 NOVEMBER 2007 MAUREPAS, 1:10 TO 3:00 P.M.

Session 3pBB

Biomedical Ultrasound/Bioresponse to Vibration: Biomedical Applications of Acoustic Radiation Force

Mostafa Fatemi, Chair
Mayo Clinic, Dept. of Physiology and Biophysics, Rochester, MN 55905

Chair's Introduction—1:10

Invited Papers

1:15

3pBB1. Modulated ultrasound and multifrequency radiation force. Matthew W. Urban, Mostafa Fatemi, and James F. Greenleaf (Dept. of Physiol. and Biomed. Eng., Mayo Clinic College of Medicine, 200 First St., SW, Rochester, MN 55905)

Modulating ultrasound can create dynamic ultrasound radiation force to induce local tissue vibration. Using different types of modulating signals produces radiation force with weighted multifrequency components. Modulation signals with frequency f_r produce force with components at multiples of f_r. Both amplitude modulation (AM) and double sideband suppressed carrier (DSB-SC) amplitude modulation were explored. Different waveforms were investigated such as sine, square, triangle, and sawtooth for modulating continuous wave ultrasound. A 3.0 MHz transducer produced ultrasound which was measured with a needle hydrophone and used to derive and analyze the spectrum of the radiation force function. The modulated ultrasound was used to vibrate a steel sphere in a gelatin phantom, and the motion was measured using a laser vibrometer. The measured spectra were used to find the frequency response of the sphere. Different modulating signals were used to produce multifrequency radiation force. The sphere frequency response was recovered using different modulating signals. Weighted multifrequency radiation force was produced by modulating ultrasound with different signals. Application of multifrequency radiation force can be used to measure the frequency response of objects or tissue. [This work was supported in part by grants EB002167, EB002640, and CA091956 from NIH.]

1:40

3pBB2. Negative radiation forces on spheres illuminated by Bessel beams: Modeling using finite elements. David B. Thiessen and Philip L. Marston (Phys. and Astron. Dept., Washington State Univ., Pullman, WA 99164-2814, thiessen@wsu.edu)

An analytical solution for the scattering of an acoustic Bessel beam by a sphere centered on the beam [P. L. Marston, J. Acoust. Soc. Am. **121**, 753–758 (2007)] has made it possible to explore the way the acoustic radiation force on elastic and fluid spheres depends on beam and material parameters. Situations have been previously noted where, even in the absence of absorption, the radiation force of the beam on the sphere is opposite the direction of beam propagation [P. L. Marston, J. Acoust. Soc. Am. **120**, 3518–3524 (2006); J. Acoust. Soc. Am. **121**, 3109 (2007)]. In the case of solid spheres, the interpretation is simplified using an analysis of the scattering [P. L. Marston, J. Acoust. Soc. Am. **122**, 247–252 (2007)]. In the present research, the finite-element method (FEM) is used to evaluate the total acoustic field in the region near the sphere. This makes it possible to evaluate the radiation force from numerical integration of an appropriate projection of the Brillouin radiation stress tensor. The result agrees with analytical results for plane wave and Bessel beam illumination. The FEM result predicts negative radiation forces for appropriate beam, material, and frequency parameters. [Supported in part by NASA.]

2:05

3pBB3. Vibro-acoustography of thyroid. Azra Allizad, Farid G. Mitri, Randall R. Kinnick, James F. Greenleaf, and Mostafa Fatemi (Mayo Clinic College of Medicine, Rochester, MN 55905, aza@mayo.edu)

Thyroid imaging with conventional ultrasound scanners does not always lead to conclusive results. Furthermore, current ultrasound imaging technology has difficulty in detecting important microcalcifications. Ambiguities in ultrasound and other thyroid imaging methods lead to a large number of unnecessary biopsies. For these reasons, alternative imaging methods capable of detecting both microcalcifications and lesions in thyroid are of great interest. This paper describes recent results on imaging human thyroids with vibro-acoustography. Experiments were conducted on eight excised human thyroids from autopsy. The specimens were each imbedded in a block of gel to simulate the surrounding tissue and to facilitate scanning. Three types of images were acquired from each specimen: X-ray, B-mode ultrasound, and vibro-acoustography. Anatomical details seen in the images were correlated and image quality was compared. X-ray images displayed calcifications with high contrast, but did not show anatomical details of soft tissue. B-mode displayed some soft tissue structures; however, the speckle masked the details and reduced the contrast. Vibro-acoustography displayed calcifications, anatomical details, and some nodules where they existed. Furthermore, vibro-acoustography images displayed tissue structures with high contrast and free from speckle. It is concluded that vibro-acoustography may be a suitable technique for thyroid imaging.

Contributed Papers

2:30

3pBB4. Acoustic radiation force biorheology. Marko Orescanin (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N. Mathews Ave., Urbana, IL 61801), David Mahr, Sureshkumar Kalyanam, and Michael Insana (Univ. of Illinois at Urbana-Champaign, Urbana, IL 61801)

Acoustic radiation force techniques are being developed to image rheological properties of engineered tissues and cell cultures. An embedded sphere couples an acoustic source field to the medium, while induced deformations are imaged dynamically via Doppler methods. We conducted three experiments. First, we modeled the response to a step force as a second-order differential equation. Fitting Doppler data to the model, we estimated shear modulus (G) and coefficient of viscosity (ν) for a broad range of tissue-like hydrogels. Measuring a 2% hydrogel over 8d, we estimated $G=450-800$ Pa, $\nu=0.26-0.4$ Pa s, in close agreement with measurements using a cone-plate viscometer. In the second experiment, we applied harmonic acoustic radiation force stimuli and measured the complex modulus of the gels over a very broad bandwidth. The advantage obtained is significantly higher SNR. Finally, spatial variations in gel rheological properties were mapped from shear waves radiating from the harmonically-driven embedded sphere. The wavelength of the shear wave indicates the shear modulus. These three methods describe spatiotemporal variations in scaffolds designed for tissue engineering and cancer cell cultures, all without physically touching the samples. Acoustic radiation force rheometry is being developed as a tool for basic biological research.

2:45

3pBB5. Estimation of tissue stiffness with the surface wave generated by ultrasound. Xiaoming Zhang, Matthew W. Urban, Randall R. Kinnick, and James F. Greenleaf (Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, Zhang.xiaoming@mayo.edu)

The mechanical response of tissues to external forces has gained considerable interest in medical diagnosis. Typically, a force is needed to produce displacement in the tissue for estimation of the tissue's elasticity. One approach is to apply an external force on the body's surface. Another emerging technique is to generate a localized force inside the tissue with ultrasound. Shear waves can be used in bulky tissues for estimating the tissue's elasticity. The bending wave can be generated and measured in vessels for noninvasive estimation of a vessel's elasticity. We have found that the Rayleigh surface wave can be generated on the tissue's surface, [X. Zhang et al., Proc. International Congress on Ultrasonics, Vienna, April 9–13, 2007], [X. Zhang et al., IEEE Trans. Medical Imaging, **26**, 843–852 (2007)]. In our method, a localized radiation force of ultrasound is remotely and noninvasively applied inside the tissue. This force can generate the shear wave as well as compression wave inside the tissue. However, only the surface wave can exist on the body' surface. The surface wave can be used to estimate and image the stiffness of tissue. Some results of using the surface wave are presented.

THURSDAY AFTERNOON, 29 NOVEMBER 2007 GRAND BALLROOM E, 2:20 TO 3:15 P.M.

Session 3pED

Education in Acoustics: Acoustics Education Prize Lecture

Mark F. Hamilton, Chair
Univ. of Texas at Austin, Dept. of Mechanical Engineering, Austin, TX 78712-0292

Chair's Introduction—2:20

Invited Paper

2:25

3pED1. Songs my students sang to me. David T. Blackstock (Appl. Res. Labs., Univ. of Texas, P.O. Box 8029, Austin, TX 78713-8029, and M.E. Dept., UT Austin, Austin, TX 78712-0292)

Does the professor teach his/her students? Or do they teach the professor? While the answer to both questions is probably a qualified yes, in looking back, I see that what I know now is largely what they taught me. After a review of the research areas in which my students and I have worked, a few examples are highlighted that show that what I had expected is not how things turned out.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

James P. Chambers, Chair
National Center for Physical Acoustics, Univ. of Mississippi, 1 Coliseum Dr., University, MS 38677

Chair's Introduction—1:05

Invited Papers

1:10

3pID1. Speech-in-noise perception and recognition. Ann R. Bradlow (Dept. of Linguist., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, abradlow@northwestern.edu) and Carol Espy-Wilson (Univ. of Maryland, College Park, MD 20742)

Recent speech research has established that, for humans, speech-in-noise and speech-in-quiet perception differ along numerous dimensions that span levels of signal encoding and linguistic representation. Listeners place more or less weight on specific acoustic cues, draw more or less on signal-independent, contextual information, and are more or less distracted by lexical neighbors depending on masker type and level. Moreover, noise has different effects at different levels for different listener populations. Machine recognition of noisy speech is well below (at least by an order of magnitude) human recognition of noisy speech. This difference is true for both additive noise and convolutive noise. Researchers have focused on the problem of machine recognition of noisy speech in several ways: Developing robust features, training systems on noisy speech, or developing speech enhancement algorithms to clean up the noisy speech signal before performing recognition. A remaining challenge for understanding how humans do and how computers should handle speech-in-noise is to develop a conceptual framework that goes beyond the division of masking effects into peripheral, energetic versus central, informational.

1:30

3pID2. Speech privacy in healthcare facilities. Gregory C. Tocci and David Sykes (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776, gtocci@cavtocci.com)

In response to the growing recognition in the U.S. that personal privacy requires more protection, two federal regulations were enacted. These are the Health Insurance Portability and Accountability Act of 1996 (HIPAA) and the Gramm-Leach-Bliley Act of 1999 (GLBA). The first includes provisions for protecting patient healthcare information and the second for protecting personal financial records information. These focus on data privacy and only obliquely refer to speech privacy; however, the enforcement guidelines for HIPAA specifically require acceptable speech privacy consistent with the intention of HIPAA. The definition of speech privacy, how to evaluate it, and what constitutes acceptable speech privacy are not addressed in these Acts nor in enforcement guidelines. To provide much needed guidance on this, ANSI S12 Working Group 44 Speech Privacy was formed in 2006 to develop standards for speech privacy. The efforts of the working group are to be discussed. In networking with the healthcare building design profession, it became evident that the American Institute of Architects (AIA) Hospital Design Manual was in great need of general acoustical, as well as speech privacy guidelines. This presentation will discuss this and the many on-going efforts in the area of healthcare acoustics.

1:50

3pID3. Noise diagnosis using nearfield acoustical holography. Sean Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202)

This presentation focuses on diagnosis of air- and structure-borne sounds by using nearfield acoustical holography (NAH) with an emphasis on applications of Helmholtz equation least-squares method for various industrial projects. The presentation is structured with strong consideration towards the end result, which is the visualization and quantification of the acoustic field and ranking of relative contributions from individual noise sources. It is emphasized that, in general, this approach does not deal with actual noise control but rather with identifying the areas where such effort may yield the most cost-effective results. Comparison of NAH-based diagnosis with other techniques such as transfer path analysis and an intensity probe is presented, and their relative advantages and disadvantages are discussed. Specific examples of identifications of sound transmission paths into a passenger vehicle compartment, fuselage of an aircraft, and disk brake squeals are presented. The reconstructed acoustic quantities, such as the acoustic pressures and normal surface velocities, are compared with the benchmark results.

Session 3pSC

Speech Communication: Vowels, Vowel Tract Modeling, and Applications (Poster Session)

Catherine L. Rogers, Chair

Univ. of South Florida, Communication Science and Disorders, Tampa, FL 33620-8150

Contributed Papers

All posters will be on display from 1:00 p.m. to 3:20 p.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:10 p.m. and contributors of even-numbered papers will be at their posters from 2:10 p.m. to 3:20 p.m.

3pSC1. Speech-to-music ratio estimation using wavelets and hidden Markov models. Brett Smolenski (Res. Assoc. for Defense Conversion, 2433 Forest Ln., Marcy, NY 13403, Brett.Smolenski.ctr@rl.af.mil)

In this paper, a system capable of estimating the long-term, on the order of an utterance, speech-to-music energy ratio (SMR) is developed. The approach uses frame-based Teager energy values at the output of a seven-band wavelet decomposition as features. The features are then modeled using a two-state hidden Markov model (HMM), with each state producing observations having a 64-component Gaussian mixture. Using this approach, estimation of the long-term SMR with a standard error of less then 5% was obtained. In addition, accurate classification of music and speech with these models was still possible when both music and speech were simultaneously present, assuming one had a higher energy than the other.

3pSC2. Effect of level of presentation on scaled speech intelligibility of speakers with dysarthria. Yunjung Kim, Gary Weismer, Raymond D. Kent (Waisman Ctr., Univ. of Wisconsin-Madison, 1975 Willow Dr., Madison, WI 53706), and Joseph R. Duffy (Mayo Clinic, Rochester, MN)

A considerable amount of effort has been made to seek the acoustic correlates of reduced speech intelligibility in dysarthria, although some acoustic variables are under debate: For example, articulation rate, F2 slope, and vowel space have been frequently discussed in previous studies as acoustic predictors of speech intelligibility of dysarthria [Kim (2007)]. One variable that is not well understood, with respect to its effect on speech intelligibility in dysarthria, is level of presentation. This includes level generated by a speaker as well as level of presentation after the speaker has recorded utterances. Clearly, variations in level of presentation of recorded utterances will have some effect when an intelligibility test requires some segmental analysis (as in a minimal pairs test), but the potential effect when speech intelligibility is scaled is not so clear. This is an important question because of the frequent use of scaled intelligibility in dysarthria research. In this presentation, we will report data on how scaled speech intelligibility is affected by level of presentation of speech samples produced by speakers with various types and severity of dysarthria. [Work supported by NIH DC00319.]

3pSC3. Effects of clear speech on duration and fundamental frequency of vowels produced by monolingual and bilingual talkers. Catherine L. Rogers, Michelle Bianchi, Stefan A. Frisch, and Jean C. Krause (Dept. of Commun. Sci. & Dis., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620)

Prosodic characteristics of vowels produced by monolingual and bilingual talkers were investigated. Ten monolingual, 15 early Spanish-English bilingual (age of onset of immersion of age 12 or earlier), and ten late Spanish-English bilingual (age of onset of immersion of age 15 or later) talkers produced the target words "bead, bid, bayed, bed, bad," and "bod" in conversational and clear speech styles. Vowel duration was computed, and F0 measurements were made at 20%, 50%, and 80% of the vowel duration. A significant group by style by vowel interaction showed that monolingual and early bilingual talkers enhanced inherent duration differences between target vowels by lengthening long vowels significantly more than short vowels in clear speech. The vowels of the late bilingual talkers, by contrast, became more alike in duration in clear than in conversational speech. The monolingual talkers showed a falling F0 pattern from 20% to 80% of the vowel duration in both styles; the late bilingual talkers showed a flat or rising F0 pattern in both styles; and the early bilingual talkers showed a flat or rising pattern in conversational speech, but a falling pattern in clear speech. [Work supported by NIH-NIDCD #5R03DC005561.]

3pSC4. Tonal coarticulation in Yoruba: Locus equation analysis. Augustine Agwuele (Texas State Univ., San Marcos, 601 Univ. Dr., San Marcos, TX 78666)

This study examines the acoustic coarticulatory effects of tone on VCV sequences of Yoruba, when each vowel of the VCV sequences bears a different tone. The first aim is to understand how the presence of tone affects CV coarticulation in the traditional locus equations parlance as used by Krull (1987). This marks the first application of the locus equation metric to a tonal language in the study of CV coarticulation. The second aim is to determine how the stop consonant in VCV sequences is affected by prosodic overlay, independently of the vowels altered location in articulatory/acoustic space. In order to achieve the second objective, the modified locus equation regression metric that was used by [Lindblom et al. J. Acoust. Soc. Am. (2007)] to dissociate vowel context effects from rate-induced effects on consonantal F2 onsets are applied. Similar to the findings of Lindblom *et al.* (2007), the analyses document separate effect for F2 locus relative to F2 nucleus.

3pSC5. Speeded discrimination of American vowels by experienced Russian learners of English. Yana D. Gilichinskaya, Franzo Law II, and Winifred Strange (Speech Acoust. and Percept. Lab., City Univ. of New York—Grad. Ctr., 365 Fifth Ave., New York, NY 10016-4309)

This study is part of a project examining how L2 learners whose native languages have small vowel inventories (Japanese, Russian, Spanish) perceive American English (AE) vowels. Experienced Russian (RU) L2 learners' discrimination was evaluated in a speeded ABX task that included nil Experimental contrasts among adjacent height pairs and front/back pairs; four nonadjacent height pairs served as Controls. An earlier study of Japanese L2 learners indicated that AE contrasts with spectrum

and duration differences (S+D) [iː/ɪ, æː/[g\], uː/ʊ, ɑː/ʌ] were discriminated more rapidly, relative to Control contrasts [iː/[g\], ɪ/æ, uː/ʌ, ʊ/ɑː], than were contrasts that differed only in spectral structure (S-Only) [ɪ/[g\], ʊ/ʌ, [g\]/ʌ, æː/ɑː]. Since vowel duration is not contrastive in Russian, it was hypothesized that discrimination of S-Only and S+D contrasts might be equally slow, relative to Controls, if RU listeners did not attend to duration differences. Preliminary findings confirm that reaction time (RT) difference scores (relative to Controls) were not different for S-Only and S+D contrasts. Relative RTs on both contrast types were slower than for AE listeners, whose RTs were about the same for S-Only and S+D pairs. These findings suggest that speeded discrimination is a sensitive measure of continuing perceptual difficulties of L2 learners. [Work supported by NSF.]

3pSC6. Perceptual similarity of American English and Japanese vowels for native speakers of American English and Japanese. Takeshi Nozawa (Program in Lang. Education, Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu Shiga 525-8577, Japan) and Elaina M. Frieda (Auburn Univ., Auburn, AL 36849)

Native speakers of American English and Japanese chose American English vowels that best represented Japanese vowels uttered in two different consonantal contexts. The two groups of subjects' responses differed noticeably, showing the effects of linguistic experience. The Japanese subjects tended to match Japanese long two-mora vowels with long, tense vowels and short one-mora vowels with short lax vowels, but the English subjects were relatively unaffected by durational differences. The Japanese subjects matched /a/ and /e/ to /æ/ and /[g\]/, respectively, but the English subjects matched /a/ with /ɑ/ or /ʌ/, and /e/ with /ɪ/ as well as /[g\]/. The Japanese subjects matched /o/ with /ɑ/, while the American subjects matched it with /oʊ/. The Japanese subjects took part in two other experiments. They equated American English vowels to Japanese vowel categories and they also identified American English vowels. They equated English /ɑ/ to Japanese /a/, and /ɑ/ was often misidentified as /æ/. The results of the three experiments show that there are discrepancies between how an American English vowel is mapped into Japanese vowel categories and how the Japanese subjects expect it to sound, which may result in the poor identification of American English vowels.

3pSC7. Vowel intrinsic structure involved in continuous speech in articulatory space. Xugang Lu, Jianwu Dang, and Satoru Fujita (Japan Adv. Inst. of Science and Technol., 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan, xugang@jaist.ac.jp)

The relation of the speech production and perception was described using the similarity of acoustic (F1-F2) space and articulatory (vowel-triangle) space for isolated vowels. However, such a relation has not been found in continuous speech. This study attempts to find out the intrinsic structure of vowels involved in continuous speech by focusing on the whole vowel structure involved in the articulation, by keeping the neighboring topological relationship of the data set. A nonlinear dimension reduction method, Laplacian eigenmap, is adopted to realize this mission. Japanese vowels were extracted from an articulatory data set that was obtained using the electromagnetic articulography system for continuous utterances. When the articulatory data were compressed from 14 dimensions to three dimensions, a nonhomogeneous structure emerged from the vowels. The vowels were distinctively categorized based on their similarity relationship revealed by their distribution on the structure along the three dimensions. For the three dimensions, the first dimension corresponds to the height of the tongue, and the second one relates to the lip rounding. The third dimension is concerned with the articulatory place along the vocal tract. The same method was also used to extract the intrinsic structure of the corresponding acoustic data set.

3pSC8. Acoustic articulatory evidence for quantal vowel categories: The feature [low]. Youngsook Jung (Speech and Hearing Bioscience and Technol., Harvard-MIT Div. of Health Sci. and Technol., Cambridge, MA 02139) and Kenneth N. Stevens (MIT, Cambridge, MA 02139)

The goal of this study is to determine whether acoustic coupling between the first subglottal resonance F1sub (about 600 Hz) and the F1 frequency for vowels creates a region near 600 Hz in which the F1 prominence shows an irregularity. Such a finding would provide evidence for a defining quantal articulatory-acoustic relation for the distinctive feature [low]. The time course of F1 in relation to F1sub was examined for certain diphthongs and several monophthongs produced by a number of speakers of English using Chi's data [X. Chi and M. Sonderegger, J. Acoust. Soc. Am. **115**, 2540–2550 (2004)]. For the diphthongs, a discontinuity in F1 or a dip in amplitude of the F1 prominence was observed as it passed through F1sub, while for the monophthongs, F1 was usually above F1sub for [+low] vowels and below F1sub for [-low] vowels. A preliminary further study of data from the literature on F1 for vowels from various languages showed that the boundary between F1 values of [+low] vowels and those of [-low] vowels agrees with the average value of F1sub obtained from the laboratory study with English. [Supported by NIH Grant No. DC00075.]

3pSC9. Incorporating vocalic segment memories in automatic dialect identification. David M. Rojas (Linguist. Dept., Indiana Univ., MM 322, 1021 E. Third St., Bloomington, IN 47405, drojas@indiana.edu)

Systematic pronunciation differences among speakers of regional varieties of U.S. English are recognizable to other native speakers to varying degrees. This has often been demonstrated through experiments wherein listeners were asked to match a talker to his or her dialect region. Machines have also been able to identify the regional origin of a speaker to some degree, although attempts to this end have typically not been as successful as efforts to automatically identify the language of a speaker. In order to refine the dialect discrimination ability of a machine, this paper draws methodological inspiration from the area of musical artist classification, and from linguistic notions that vowels contribute more heavily than consonants to regional differences. Using this insight, an automatic dialect identification system is developed that first recognizes the more vowel-like slices of the signal, and then updates a vocalic segment memory component with Mel-frequency cepstral coefficient, formant, and pitch information from the current frame. Besides providing a means to analyze MFCC and formant trajectories, the segment memory enriches the representation of vocalic events by allowing the system to explicitly model prosodic aspects such as duration and tilt.

3pSC10. The application of a psychophysical difference metric to perceptual similarity judgments in vowels. James Harnsberger, Rahul Shrivastav (Inst. for Adv. Study of the Commun. Proc., Univ. of Florida, Gainesville, FL 32611), and Mark Skowronski (Univ. of Western Ontario, London, ON N6A 5B7, Canada)

Models of cross-language speech perception have had limited success in predicting the discriminability or perceptual similarity of non-native contrasts. These failures may be attributed partly to an inability to quantify the phonetic differences between non-native speech sounds. This study attempted to quantify such gross psychophysical differences between speech sounds, specifically by utilizing dynamic time warping (DTW) on human factor cepstral coefficients to compare the spectrum of the entire length of the speech sounds in question. This technique has been successfully applied to account for the discriminability of different non-native consonant contrasts [Harnsberger, J. D., Shrivastav, R., and Skowronski, M.; J. Acous. Soc. Am. 117, 2460, 2005]. This study extends this work to perceptual similarity judgments of vowels. Specifically, twenty native speakers of English were presented with all possible pairings of ten vowels produced by two speakers of English. Subjects were asked to rate their similarity on a seven point scale. The resulting similarity scores were then compared with the output matrix of the DTW psychophysical difference

metric for the same stimulus materials. The results showed a significant correlation (r=.60**) between the two measures, demonstrating the efficacy of the metric with a greater range of stimulus types and tasks.

3pSC11. Effects of several consonant environments on vowel formants. Michael Kiefte (School of Human Commun. Disord., Dalhousie Univ., Halifax, NS B3H 1R2 Canada)

A large body of evidence has shown that relative change in spectral pattern possesses some invariant properties for vowels across speakers in /hVd/ environment. While formant frequency plots from vowel steady-states result in categories with large overlap, better separation of vowel categories is obtained using onset and offset formant frequencies [e.g., Hillenbrand *et al.*, J. Acoust. Soc. Am. **97**, 3099–3111 (1995)]. To a large extent, this invariance was also shown in other /CVC/ contexts using all combinations of /h,b,d,g,p,t,k/ [Hillenbrand *et al.*, J. Acoust. Soc. Am. **109**, 748–763 (2001)]. The present study explored spectral change in vowels produced in environments where larger differences might be expected across contexts. Nineteen men and 39 women were asked to produce fourteen Canadian English vowels in /hV/, /hVd/, /hVt/, /hVl/, /hVr/, /hVnd/, /hVg/, and /dVd/. Subjects read standard English orthographic representations of the target words which were embedded both in a sentence which indicated the rhyme of the target word—e.g., "Swooned rhymes with hoond" as well as in a sentence which indicated the pronunciation of onset and vowel—e.g., "Hood sounds like hoog." Formant frequencies were measured and tracked for each token and differences across consonant contexts were analyzed. [Work supported by SSHRC.]

3pSC12. Speaker normalization using cortical strip maps: A neural model for steady state vowel identification. Heather Ames and Stephen Grossberg (Dept. of Cognit. and Neural Systems and CELEST, Boston Univ., 677 Beacon St, Boston, MA 02215)

Auditory signals of speech are speaker-dependent, but representations of language meaning are much more speaker-independent. Such a speaker normalization transformation enables speech to be learned and understood from different speakers. A neural model is presented that performs speaker normalization to generate a pitch-independent representation of speech sounds, while also preserving information about speaker identity. This speaker-invariant representation is categorized into unitized speech items, which input to sequential working memories whose distributed patterns can be categorized, or chunked, into syllable and word representations. The proposed model circuits fit into an emerging theory of auditory streaming and speech categorization in which auditory streaming and speaker normalization both use similar neural designs; namely, multiple cortical strip map representations of auditory signals. This design homology may clarify how speaker normalization circuits evolved from more primitive streaming mechanisms. Simulations with synthesized steady-state vowels from the Peterson and Barney (1952) vowel database achieve accuracy rates similar to those achieved by human listeners. These results are compared to behavioral data and other speaker normalization models. [Work supported in part by NSF and ONR.]

3pSC13. Vowel space and formant dynamics of vowels produced by monolingual and bilingual talkers in conversational and clear speech styles. Michelle Bianchi, Catherine L. Rogers, Stefan A. Frisch, and Jean C. Krause (Dept. of Commun. Sci. & Dis., Univ. of South Florida, 4202 E. Fowler Ave., PCD 1017, Tampa, FL 33620)

Formant frequency characteristics of vowels produced by monolingual and bilingual talkers were compared. Ten monolingual, 15 early Spanish-English bilingual (age of onset of immersion is 12 or earlier), and ten late Spanish-English bilingual (age of onset of immersion of age 15 or later) talkers produced the target words "bead, bid, bayed, bed, bad," and "bod" in conversational and clear speech styles. Measurements of F1 and F2 were made at 20%, 50%, and 80% of vowel duration. Significant group effects for F1 and F2 at 50% of vowel duration showed similar locations and distances between the vowels for the monolingual and early bilingual talkers, except that F2 values were significantly higher for the early bilingual than for the monolingual talkers for the vowels in the target words "bead, bid," and "bed." Smaller between-vowel distances were found for both F1 and F2 for the late bilingual talkers, especially for the vowels in the target words "bead, bid, bayed," and "bed." Changes observed in clear speech were relatively modest for all three groups. Comparisons between formant dynamic vectors and angles across groups and style and between talkers showing large versus small degrees of clear speech intelligibility benefit will be presented. [Work supported by NIH-NIDCD No. 5R03DC005561.]

3pSC14. Robust unsupervised extraction of vocal tract variables from midsagittal real-time magnetic resonance image sequences using region segmentation. Erik Bresch and Shrikanth Narayanan (Univ. of Southern California, 3740 McClintock Ave., Rm. EEB400, Los Angeles, CA 90089)

The tracking of deformable objects in image sequences has been a topic of intensive research for many years, and many application-specific solutions have been proposed. In this work, we describe a method that was developed to robustly track the tissue structures of the human vocal tract in midsagittal, real-time magnetic resonance (MR) images. The goal of the algorithm is to fully automatically extract the vocal tract outline, the position of the articulators, and the tract variables to facilitate the study of the shaping of the vocal tract during speech production. The algorithm is unsupervised and requires only a one-time initialization step for a particular subject. Importantly, the tracking algorithm operates on the spatial frequency domain representation of the underlying images, and it is hence specifically fit to the data produced in the MR imaging process. The proposed method carries out a multiregion segmentation of the individual MR images using an anatomically informed model of the vocal tract whose fit to the observed image data is hierarchically optimized using an anatomically informed gradient descent procedure. The mathematical key components of the algorithm are the closed-form solution of the two-dimensional Fourier transform of a polygonal shape function and the design of alternative gradient descent flows for the iterative solution of an overdetermined nonlinear least squares optimization problem. Various examples of segmented real-time MR images and a summary of open challenges will be presented. [Work supported by NIH.]

3pSC15. Volumetric MRI acquisition and processing. Inês Carbone (Dept. Electrons, Telecommunications and Informatics/IEETA, Univ. of Aveiro, Campus Universitário de Santiago, 3810-193 Aveiro, Portugal), Paula Martins, Augusto Silva, and António Teixeira (Univ. of Aveiro, 3810-193 Aveiro, Portugal)

To provide fast and potentially more accurate three-dimensional anatomic information on European Portuguese (EP) sounds, a direct acquisition of volumetric MRI information was attempted, through a 3-D spoiled fast gradient echo sequence. The viability of the used MRI acquisition protocol relies on the possibility of obtaining consistent information. For that, two different image processing approaches were attempted: First, a so-called 2.5-D method consisting in a first segmentation on the sagittal plane, multiplanar reslicing perpendicularly to the sound propagation in the vocal tract, and new segmentation in the 45 planes obtained; second, a direct 3-D level set segmentation. A method was applied to a comprehensive single speaker corpus contemplating: Oral and nasal vowels, nasal consonants, unvoiced fricatives, and laterals. Method advantages are: Reconstruction of slices with good detail in any direction, useful to obtain slices perfectly orthogonal to the vocal tract centerline; higher SNR when compared with 2-D imaging; easier 3-D visualization; and acquisition two times faster, with positive impact in the amount of material acquired in one session with less subject effort. 3-D information was especially important for the study of nasal [P. Martins et al., InterSpeech 2007, accepted] and lateral sounds, some of them with specific characteristics for EP. [Work supported by FCT, Portuguese Research Agency, by Project HERON POSC/PLP/57680/2004.]

3pSC16. MRI study of coarticulation in European Portuguese. Paula Martins (Escola Superior de Sade da Universidade de Aveiro, Univ. of Aveiro, Campus Universitário de Santiago, 3810-193 Aveiro, Portugal), Inês Carbone, Augusto Silva, and António Teixeira (Univ. of Aveiro, 3801 193 Aveiro, Portugal)

For this study, a small part of a recently acquired MRI database for European Portuguese (EP) was used. The sounds studied were the 12 EP voiced and unvoiced fricatives and stop consonants, produced in a symmetric VCV context. The vowels chosen were the cardinal vowels [i, a, u]. The speaker sustained the fricative sounds and maintained stop articulation during the 5.6 s of the acquisition time (T1 TSE sequence). The contours were obtained using the seeded region growing method. The influence of the seed placement was evaluated through the generation of 100 contours, with random seeds, for each sound. It was possible to verify, for the EP, some facts already reported by other authors relative to coarticulation in other languages. In general, EP stops are less resistant to coarticulatory effects than fricatives. The EP sounds presenting the highest resistance to coarticulation are the unvoiced postalveolar fricative [ʃ] and corresponding voiced fricative. As an example of other relevant results, for stops [t], [d] and fricative [s] there is no significant effect of the vowel in the region tongue blade region, being the influence evident in the production of the stops [k] and [g]. [Work supported by FCT, Portuguese Research Agency, by Project HERON POSC/PLP/57680/2004.]

3pSC17. Anatomic development of the vocal tract during the first two decades of life: Evidence on prepubertal sexual dimorphism from MRI and CT studies. Houri K. Vorperian, E. Michael Schimek (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave. # 430, Madison, WI 53705), Shubing Wang, Moo K. Chung, Ray D. Kent, Andrew J. Ziegert, and Lindell R. Gentry (Univ. of Wisconsin-Madison, Madison, WI)

The growth of the vocal tract (VT) is known to be nonuniform insofar as there are regional variations in anatomic maturation. This study presents quantitative anatomic data on the developing VT from 604 imaging studies (277 male; 327 female) between birth and 20 years. Data analyses include detailed assessment on the growth of the oral (anterior or horizontal) and pharyngeal (posterior or vertical) regions of the VT for both sexes. The oral region of the VT was segmented into lip-thickness, anterior-cavity-length, and orophayrngeal-width; and the pharyngeal region of the VT into posterior-cavity-length, and nasopharyngeal-length. Findings from all variables/segments indicate differences in growth trend, rate, and type (somatic versus neural) between males and females. However, it appears that prepubertal sex differences at specific age ranges are masked by overall growth rate differences between males and females. Comparing males versus females, using a limited age range of 60 months, unveiled prepubertal sexual dimorphism in the oral/horizontal region where differences appear more pronounced in the oropharyngeal-width segment. Such novel anatomic findings indicate one possible source for noted sex differences in formant frequencies before age 10, where there is no sexual dimorphism in VT length. [Work supported by grants NIH-NIDCD R03-DC4362, R01-DC006282, and NIH-NICHHDP30-HK03352.]

3pSC18. Production facilitates lexical acquisition in young children. Tania Zamuner, Paula Fikkert (Ctr. for Lang. Studies, Radboud Univ. Nijmegen, 6500 HD Nijmegen, The Netherlands, tzamuner@psych.ubc.ca), and Bryan Gick (UBC, Vancouver BC, V6T 1Z1, Canada)

Recent research in language acquisition argues that production plays a unique role in the acquisition of phonological and lexical representations [Fikkert. In press, LabPhonX]. In this research, we investigate the effect of production in early lexical acquisition. Twenty Dutch-learning children participated in a word-learning task, where children were taught and asked to imitate non-words. Children were then tested on their recall of the non-words, and their receptive knowledge of the non-words. Children's performance on the two tasks was compared to whether they imitated the non-words during training, the number of times they imitated the non-words, and the accuracy of their imitations. While no effect of production during training was found on children's receptive knowledge of the non-words, production during training was associated with children's ability to recall the non-words. Specifically, children who produced the non-words during training were more likely to recall the non-words at test. No relationship was found between the number of times children produced the non-words during training, or the accuracy of their imitations. These results indicate that production enhances early lexical acquisition, and suggests a unique role for articulation in the formation of phonological and lexical representations.

3pSC19. How acoustic cues in infant-directed speech facilitate 19-month-olds' word recognition: Evidence from time course analysis. Jae Yung Song (Dept. of Cognit. and Linguistic Sci., Brown Univ., Box 1978, Providence, RI 02912, Jae_Yung_Song@brown.edu)

This study investigated how the acoustic characteristics of infant-directed speech (IDS) facilitate infants' recognition of familiar words. The individual roles of three typical characteristics of IDS were examined: Exaggerated pitch range, slow speech rate, and hyperarticulated vowels. Using the intermodal preferential looking procedure, 34 19-month-olds were presented with 12 stimulus sentences (Where is the [target]?); six were spoken in typical IDS style and the other six were digitally altered to remove one of the three IDS characteristics. Infants' looks to target and distracter were coded frame by frame to examine the time course of word recognition. Results showed that infants' ability to recognize words was affected by speech rate. Total looking time to the target was greater ($t(11)=3.337$, $p=0.007$), latency of the first look was shorter ($t(71)=-1.919$, $p=0.059$), and the accuracy score was higher overall when listening to typical IDS as compared to accelerated IDS. In contrast, infants showed no difference in performance when pitch range or the degree of hyperarticulation was modified. This suggests that slow speech rate enhances infants' ability to recognize words more efficiently and reliably during the stage of rapid vocabulary development. [Work supported by NSF BCS-0544127, NIH R01MH60922.]

3pSC20. Investigation of the effect of low frequency components of sound to speech intelligibility. Mokhtar Harun, Mohamad Ngasri Dimon, Siti Zaleha Abdul Hamid, and Hamim Nasoha (Acoust. Res. Lab., Faculty of Elec. Eng., Universiti Teknologi Malaysia Skudai, 81310 Skudai Johor, Malaysia)

Sound attenuation depends on source-to-listener distance, the finishes, and the shape of the room. The amount of sound attenuation at different frequencies influences the overall speech intelligibility in a room. The sound at high frequencies, especially at 1 kHz and 2 kHz, contribute the most for the articulation of consonants, and thus affects speech intelligibility. However, it has been found previously that sound attenuation is lower and harder to achieve at lower frequencies. This would lead us to assume that reverberant sound is mostly of low frequency components. Even if the contribution of sound at low frequency is low for speech intelligibility, this paper attempts to gauge the effect of sound pollution at low frequencies to the overall speech intelligibility in the room. With fixed source-to-listener distances, the measurements were conducted in classrooms with different sizes and finishes.

3pSC21. Children's perception of American English consonants in noise. Kanae Nishi, Dawna E. Lewis, Brenda M. Hoover, Sangsook Choi, and Patricia G. Stelmachowicz (Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131)

Following the seminal study [Miller and Nicely, J. Acoust. Soc. Am. **27**(2), 338–352 (1955)], numerous studies have explored consonant confusion patterns for adult listeners. Although children are known to be less skilled than adults in many aspects of auditory abilities, little is known about how these immature abilities affect consonant perception. In the current study, 72 children (4-, 5-, 6-, 7-, 8-, and 9-year-olds) and 12 adults identified 15 American English consonants in /aCa/ nonsense syllables presented in speech-shaped noise at 0, $+5$, and $+10$ dB signal-to-noise ratios (SNRs). Overall percent-correct identification of 15 target consonants improved with age and SNR, but no SNR age interaction was observed. Further analysis of confusion patterns using indexes for transmitted information (TI) for linguistic features (manner, place, and voicing) indicated the following: (1) TI for all features increased with improvement in SNR for all age groups; (2) TI for voicing did not vary with age; (3) TI for place was significantly lower for younger children than adults. These results will be discussed in relation to the development of auditory skills. [Work supported by NIH DC004300.]

3pSC22. The contributions of global spectral and amplitude structure to speech perception by English-speaking adults and children and Mandarin-speaking adults. Susan Nittrouer and Joanna H. Lowenstein (Speech and Hearing Sci., Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, nittrouer.1@osu.edu)

Two studies examined the recognition of signals preserving only the global spectral or amplitude structure of speech by three groups of listeners: Native English-speaking adults, native English-speaking 7-year-olds, and native Mandarin-speaking adults who learned English as a second language. Stimuli consisted of sine wave (SW) replicas and amplitude-modulated (AM) noise bands derived from semantically anomalous four-word sentences (Experiment 1) and semantically rich five-word sentences (Experiment 2). In both experiments, children demonstrated recognition scores comparable to those of the English-speaking adults for the SW sentences, but scores comparable to those of the Mandarin-speaking adults for the AM sentences. In both experiments, native Mandarin-speaking adults were much worse at understanding either kind of stimulus than were the English-speaking adults. Use of linguistic constraints was similar across listener groups, diminishing the possibility that results could be attributed to differences in "top-down" effects. Results were used to support three main conclusions: (1) Global spectral and amplitude structure varies across languages; (2) native speakers use this information for speech perception; and (3) in learning a native language, children discover global spectral before global amplitude structure. [Work supported by NIDCD Grant No. DC-00633.]

3pSC23. Identification of consonant-nucleus-consonant words produced by a female child. Morgan Meredith (Dept. of Communicative Disord., UW-Madison, 1975 Willow Dr., Madison, WI 53706, mmeredith@wisc.edu)

The CNC word identification test is part of the speech test battery used to evaluate speech perception by persons fitted with cochlear implants [Luxford et al, Otolaryngology-Head and Neck Surgery, 126–127 (2001)]. This test uses ten lists of 50 relatively common, phonetically balanced English words per list [Peterson and Lehiste, J. Speech. Hear. Disorders, 27:62–70 (1962)]. The version of this test, currently standardized and used across the U.S., contains the words spoken by one male speaker. Research has shown, however, that persons with cochlear implants have more difficulty perceiving speech produced by women and children than speech produced by men [Loizou et al,. J. Acoust. Soc. Am., 103: 1141–1149 (1998)]. The ten CNC word lists were recorded as produced by an eleven-year old girl. Each word was isolated and the word ready spoken by the same girl was added at the beginning of each stimulus sound file. Ten adults with normal hearing identified these words with 98% accuracy. Identification results will be obtained from persons fitted with cochlear implants and compared to the results of the same persons when listening to the original, male voice, CNC word lists. [Work supported by a UW-Madison Hilldale Research Scholarship.]

3pSC24. Why do children attend to global structure in speech signals? Joanna H. Lowenstein and Susan Nittrouer (Speech and Hearing Sci. The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210. lowenstein.6@osu.edu)

Adults can use global structure such as that found in sine wave and vocoded signals for speech perception. Children appear to weight global spectral structure (represented at the syllabic level by formant transitions) even more than adults, prompting the suggestion that children first attend to the relatively slow vocal-tract movements that create global spectral structure rather than to the rapid and discrete gestures more closely associated with individual phonetic segments. But this interpretation hinges on an articulatory account of speech perception, which lacks general support. An alternative explanation is that children attend to formant transitions because they are found in voiced signal portions and so adhere to principles of auditory scene analysis, such as having a common fundamental. This work tested that hypothesis by using stimuli with no fundamental. Adults and children (3, 5, and 7 years) were asked to label fricative-vowel syllables with three kinds of vocalic portions: natural, sine wave, and whispered. Results showed that all listeners attended to formant transitions as much in the sine wave and whispered stimuli as in the natural, leading to rejection of the hypothesis that children prefer formant transitions because they are signal components that adhere to principles of auditory scene analysis. [Work supported by NIDCD Grant No. DC-00633.]

3pSC25. Voicing and devoicing in similar German and English word pairs by native speakers of German. Bruce L. Smith (Dept. of Commun. Sci. and Disord., Univ. of Utah, 1201 Behavioral Sci. Bldg., Salt Lake City, UT 84112), Michael Bruss (Saarland Univ., Saarbruecken, Germany), Rachel Hayes-Harb and Amy Hamilton (Univ. of Utah, Salt Lake City, UT 84112)

A number of languages manifest a pattern of word-final obstruent devoicing, i.e., words ending with underlying voiced obstruents are pronounced so as to be (largely) indistinguishable from words ending with final voiceless obstruents. One question regarding this tendency to devoice voiced obstruent targets is what occurs when native speakers of such a language learn a second language that has a word-final obstruent voicing contrast. For example, do German speakers neutralize voicing contrasts in English as they tend to do in German, or can they learn to produce voiced obstruents in Englis, despite devoicing them in German? To examine this issue, ten native speakers of German produced various phonologically-similar minimal pairs in German and English (e.g., English: Lied/light; German: Leid/leit). Acoustic measurements were made of their productions in both languages, viz., vowel duration preceding final consonants, final consonant duration, duration of voicing during final consonants, and final release burst duration; their English productions were also compared to those of native English speakers. The native German speakers tended to neutralize the voicing distinction to a greater extent when producing German words versus phonologically-similar English words, but they typically did not produce as much of a contrast in English as native English speakers.

3pSC26. Psychometric functions and psychometric characteristics of Mandarin monosyllables. Kuen-Shian Tsai, Sheunn-Tsong Young (Inst. of Biomed. Eng., Natl. Yang-Ming Univ., No. 155, Sec. 2, Linong St., Taipei City 112, Taiwan, young@bme.ym.edu.tw), Li-Hui Tseng, and Cheng-Jung Wu (Natl. Taipei College of Nursing, Taipei City 108, Taiwan)

The psychometric functions of the 700 most frequently occurring Mandarin monosyllables (whose cumulative usage was 98.38% in 1,125 distinct monosyllables) were evaluated. Twenty normal-hearing subjects were asked to hear and to repeat the 700 monosyllables which were randomly presented at the level from 0 to 55 dB HL in 5-dB step. The psychometric functions for each of the 700 monosyllables were fit with third-degree polynomials. The fitted curves were used to calculate five psychometric characteristics: (1) the threshold at 0% correct; (2) the threshold at 50% correct; (3) the instantaneous slope at 50% correct; (4) the linear slope from 20% to 80% correct; and (5) the intelligibility at the highest presentation level. The mean values of such five psychometric characteristics for 700 monosyllables are 1.1 dBHL, 12.1 dBHL, 4.5 %/dB, 4.1% %/dB, and 93.4%, respectively. With the psychometric characteristics, the 700 monosyllables can be used to create word recognition test lists for different purposes, such as constructing test lists with different homogeneity, different familiarity, or different difficulty; and the recognition performance of test lists will be predictable. Futhermore, the research results also indicated that there is no significant correlation between the psychometric characteristics and the occurring frequences of monosyllables.

THURSDAY AFTERNOON, 29 NOVEMBER 2007 GRAND BALLROOM D, 1:00 TO 3:05 P.M.

Session 3pUW

Underwater Acoustics: Target Scattering

David C. Calvo, Chair

Naval Research Laboratory, Acoustics Div., 4555 Overlook Ave., SW, Washington, DC 20375-5350

Chair's Introduction—1:00

Contributed Papers

1:05

3pUW1. Numerical solution of high-frequency acoustic scattering problems by integral equations preconditioned using pseudo-differential impedance operators. David Calvo (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375), and Paul Calamia (Rensselaer Polytechnic Inst., Troy, NY)

Exact computations of high-frequency scattering by targets or rough surfaces can be challenging because of the large number of unknowns. A compromise can be obtained, however, using approximate Dirichlet-to-Neumann surface operators that are known in pseudo-differential form and efficiently implemented using rational approximations. These operators can be used as preconditioners to obtain well-conditioned iterative algorithms to solve the Helmholtz integral equation. Using these algorithms, we compute scattering by two- and three-dimensional targets, and rough surfaces with smooth concave or convex parts with incident fields near grazing angles of interest. This case is particularly important for underwater acoustics, where sounds field graze the free surface or bottom in shallow water.

1:20

3pUW2. At-sea measurements and modeling of acoustic scattering from solid-filled elastic targets on the sea-bed. Mario Zampolli, Alessandra Tesei, and Gaetano Canepa (NATO Undersea Res. Ctr., Vle. S. Bartolomeo 400, 19126 La Spezia, Italy)

Monostatic and bistatic measurements of acoustic scattering from solid-filled fiberglass objects (a sphere and a cylinder with hemispherical endcaps) deployed proud on a sandy seabed and insonified by a rail-mounted parametric source at low frequency (roughly ka=540), are presented. The targets are thin-walled, approximately isotropic random-fiberglass shells, filled partially with an isotropic epoxy resin and partially with sea water. The targets are simple enough in shape to be treated by currently available modeling techniques, but realistic enough to give a first insight into the physics of the elastic waves supported by such material combinations. Preliminary data analysis indicates that the scattering signatures are dominated by the solid filling. The experimentally measured target responses are compared to numerical simulations obtained with a finite-element tool based on the decomposition of the field variables in a series of azimuthal Fourier modes. The experimental data were acquired in October 2006 during the EVA-06 trial off the Island of Elba, Italy.

1:35

3pUW3. Tank measurements and modeling of elastic scattering by resin-filled fiberglass spherical shells. Alessandra Tesei, Mario Zampolli, and Piero Guerrini (NATO Undersea Res. Ctr., Vle. S. Bartolomeo 400, 19126 La Spezia, Italy)

Acoustic elastic scattering measurements were conducted in a tank on 6 cm-radius fiberglass spherical shells filled partially or completely with a low-shear-speed epoxy resin. Preliminary measurements were conducted also on the void shell before filling and on a solid sphere of the same material of the filler, in order to estimate the constituent material parameters via acoustic inversion. The objects were measured in the backscatter direction, suspended at midwater, and insonified by a broadband directional transducer. From the inspection of the response of the solid-filled shell, it was possible to detect and characterize inhomogeneities of the interior (air inter-layers), the presence of which were later confirmed by CT scan and ultrasound measurements. Elastic wave analysis and modeling tools supported the physical interpretation of the measured responses. A spherical shell partially filled with epoxy resin and water was found to exhibit a strong focusing effect in the backscattering direction, for near-perpendicular incidence with respect to the planar internal water-resin interface. Spheres of this kind have the potential of being employed as sonar calibration targets.

1:50

3pUW4. Internal field in an immersed, absorbing fluid sphere excited by a plane acoustic wave. Kenneth G. Foote (Woods Hole Oceanograph. Inst., 98 Water St., Woods Hole, MA 02543)

Modeling of sonar interactions with complicated bodies whose characteristic dimensions are of the order of or greater than the acoustic wavelength generally requires use of numerical methods. For validation of as-

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sociated computer code, a particularly useful example is the internal field excited in an immersed, homogeneous, absorbing sphere by an external plane acoustic wave, since this can be expressed analytically. Here, numerical results are given for a fluid sphere with wave number-radius products spanning the range 1–10 and absorption coefficients spanning the range 0–10 dB per wavelength. The density and sound speed contrasts of the fluid sphere relative to the immersion medium are both varied over the range 0.5–2. [Work supported by NOPP through ONR award no. N000140710992.]

2:05

3pUW5. Scattering analysis accelerated by a 3-D multilevel non-uniform grid field evaluation algorithm. Yaniv Brick and Amir Boag (School of Elec. Eng., Tel Aviv Univ., Tel Aviv 69978, Israel, boag@eng.tau.ac.il)

A fast algorithm for computing the scattering cross section of arbitrary shaped large rigid bodies using an iterative method of moments solver has been presented. The main computational bottleneck of such iterative solvers stems from the need to perform at each iteration at least one matrix-vector product. If performed directly, matrix-vector multiplication, which is equivalent to field evaluation for a given source distribution, is characterized by $O(N^2)$ complexity (N being the number of unknowns). To that end, a multilevel non-uniform grid (MLNG) algorithm for 3-D fast field evaluation has been proposed, developed, and tested on representative examples of elongated, quasi-planar, and full 3-D scatterers. The algorithm relies on hierarchical domain decomposition, field evaluation on highly sparse non-uniform grids, and multilevel field aggregation through phase-compensated interpolations. Computational complexity and memory requirements of $O(N \log_{10} N)$ have been achieved by the MLNG without affecting the convergence of the iterative solver. Complexity of the MLNG is similar to that of the multilevel fast multipole algorithm [S. Ko and W. C. Chew, J. Acoust. Soc. Am. **103**, 721–734 (1998)]. The MLNG approach is inherently geometrically adaptive, provides seamless transition from the high frequency to quasi-static regime, and is quite easy to implement.

2:20

3pUW6. The Lloyd-Berry effective wave number in multiple scattering. Dalcio K. Dacol and Gregory J. Orris (Acoust. Div., Naval Res. Lab., Washington, DC 20375-5350)

The scattering of an acoustic field by an ensemble of many discrete objects is a phenomenon observed in many different contexts. In underwater acoustics, one finds examples in propagation through bubbly water, fish schools, non-homogeneous sediments (embedded shells for example), etc. The effective wave number associated with the propagation of the statistically averaged acoustic field scattered by an ensemble of many scatterers is a quantity of physical interest that is also experimentally accessible. Theoretically, this quantity can be evaluated only approximately. Typically, it is evaluated as an expansion in the average density of scatterers. In the literature, one finds two distinct expressions for the second-order contribution, one, the oldest, due to Waterman and Truell, and another by Lloyd and Berry. A new derivation of the Lloyd-Berry formula is provided using an approximation similar to the one used by Waterman and Truell and also by Foldy and Lax. This derivation should shed light on the origin of the different expressions and lend support to the contention that the Lloyd-Berry expression is the correct second-order term in the density expansion.

2:35

3pUW7. Scattering by a sphere illuminated by a Bessel beam: Finite-element modeling. David B. Thiessen and Philip L. Marston (Phys. and Astron. Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

The recent analytical solution for the scattering of an acoustic Bessel beam in water by a sphere centered on the beam [P. L. Marston, J. Acoust. Soc. Am. **121**, 753–758 (2007)] gives insight into the processes associated with excitation of target resonances and provides a benchmark for numerical approaches. For example, it is possible to suppress the excitation of certain modes of an elastic sphere by appropriate selection of the beam parameters [P. L. Marston, J. Acoust. Soc. Am. **122**, 247–252 (2007)]. The present work demonstrates that computation of the scattering for steady state illumination by a Bessel beam using the finite-element method (FEM) reproduces the analytical result for the scattering. The dependence of the mode coupling on beam parameters is also present in the FEM results. Some aspects of FEM-based scattering calculations for acoustic beams will be noted. [Work supported by ONR.]

2:50

3pUW8. Enhanced decay rate for the depth dependence of the backscattering of sound by simulated buried targets with evanescent wave illumination. Philip L. Marston, Aubrey L. Espana, Curtis F. Osterhoudt, and David B. Thiessen (Phys. and Astron. Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Sound incident, with a sufficiently small grazing angle on a smooth sediment, produces an acoustic evanescent wave in the sediment, which decays exponentially with increasing depth. A system consisting of an immiscible liquid in contact with a second liquid, has been used in the laboratory to explore the backscattering of sound by a cylinder illuminated by an evanescent wave [C. F. Osterhoudt, Ph.D. Thesis, Washington State University (2007)]. In that work the organ-pipe mode of an open-ended liquid filled metal cylinder was excited. The backscattering was found to decay with increasing depth at a rate approximately twice the spatial decay rate of the incident wave in the simulated sediment. Reciprocity predicts the decay rate is doubled in the case of localized acoustic coupling at the target. Recent supporting measurements of the backscattering, associated with a low-frequency circumferential mode of a plastic cylinder, are noted. Finite-element-based computations of the far-field low-frequency backscattering by a rigid cylinder in the simulated sediment show a decay rate approximately double that of the incident evanescent wave. [Supported by ONR.]

Plenary Session and Awards Ceremony

Gilles A. Daigle, Chair
President, Acoustical Society of America

Business Meeting of the Acoustical Society of America

Motion to approve changes to the Bylaws of the Acoustical Society of America

Presentation of Certificates to New Fellows

William A. Ahroon
Jeffrey E. Boisvert
Elizabeth A. Cohen
Dimitri M. Donskoy
Bruce E. Douglas
William T. Ellison
Ronald L. Eshelman
Steven I. Finette
Sarah Hawkins
Jean-Pierre Hermand
R. Glynn Holt
Anthony P. Lyons
Masayuki Morimoto
Valdimir E. Ostashev
Brandon D. Tinianov
Ronald A. Wagstaff

Presentation of Science Writing Awards

Science Writing Award in Acoustics for Journalists

John Geirland for ''The Sound of Silence,'' WIRED magazine, December 2006

Don Monroe for ''Why the Inner Ear is Snail Shaped,'' Physical Review Focus website, March 2006

Science Writing Award for Professionals in Acoustics

Gary S. Settles, ''High-speed imaging of shock waves, explosions, and gunshots,'' in American Scientist magazine, January/February 2006

Presentation of Acoustical Society Medal Awards

Rossing Prize in Acoustics Education to David T. Blackstock

Pioneers of Underwater Acoustics Medal to William M. Carey

Silver Medal in Engineering Acoustics to Allan J. Zuckerwar

Silver Medal in Speech Communication to Ingo R. Titze

Session 3eMU

Musical Acoustics: Science and Performance

Uwe J. Hansen, Cochair
Indiana State Univ., Physics Dept., Terre Haute, IN 47809

Tyrone M. Porter, Cochair
Boston Univ., Dept. of Aerospace and Mechanical Engineering, Boston, MA 02215

Invited Paper

7:00

3eMU1. Musical acoustics: Science and performance. Uwe J. Hansen (Indiana State Univ., Terre Haute, IN 47809) and Tyrone Porter (Boston Univ., Boston MA 02215)

ASA members, local residents and students are invited to this session featuring a local Jazz combo. The scientific basis of each instrument will be introduced and performance examples will illustrate each instrument family in turn. The evening will conclude with performances of a number of pieces by the combo.